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DATE:	**August 10, 2006**	
TO:	**Mail Box: Appeal Brief-Patents** **Examiner Elahee, MD S.**	**Group Art Unit: 2645**
COMPANY:	**United States Patent and Trademark Office**	
FACSIMILE NO:	**571-273-8300**	
FROM:	**H. Artoush Ohanian, Reg. No. 46,022**	
RE:	**Retransmission of Appeal Brief dated July 27, 2005; Title: "Destination Device Based Callee Identification"**	**Atty. Docket No.: AUS920010823US1 (129)**
SERIAL NO.:	**10/015,280**	**Customer No. 34533**
NUMBER OF PAGES:	**(Including Cover) 45**	
COMMENTS:	**Please see attached.**	

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TRANSMITTAL FORM

(to be used for all correspondence after initial filing)

Application Number	10/015,280
Filing Date	12/12/2001
First Named Inventor	Michael W. Brown
Art Unit	2645
Examiner Name	Elahee, MD S.
Total Number of Pages in This Submission: 45	Attorney Docket Number: AUS920010823US1

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  - [ ] Affidavits/declaration(s)
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- [ ] Certified Copy of Priority Document(s)
- [ ] Reply to Missing Parts/ Incomplete Application
  - [ ] Reply to Missing Parts under 37 CFR 1.52 or 1.53
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- [ ] Appeal Communication to Board of Appeals and Interferences
- [x] Appeal Communication to TC (Appeal Notice, Brief, Reply Brief)
- [ ] Proprietary Information
- [ ] Status Letter
- [x] Other Enclosure(s) (please Identify below):

Copy of Postcard; Retransmission of Appeal Brief Correspondence(2 pgs.), Copy of Appeal Brief dated 07/27/2005 (36 pgs), & 5 References

Remarks

The Commissioner is authorized to charge or credit Deposit Account No. 09-0447.

Customer No. 34533.

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Firm Name	Biggers & Ohanian, LLP		
Signature			
Printed name	H. Artoush Ohanian		
Date	August 10, 2006	Reg. No.	46,022

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Typed or printed name	Catherine Berglund	Date	August 10, 2006

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AUS920010823US1
APPEAL BRIEF

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

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In re Application of: Michael Wayne Brown, *et al.*	§	Group Art Unit: 2645
Serial No.: 10/015,280	§	Examiner: Elahee, MD S
Filed: December 12, 2001	§	Atty Docket No.: AUS920010823US1
Title: Destination Device Based Callee Identification	§	Customer No. 34533

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August 10, 2006
Date
Catherine Berglund
Catherine Berglund

RETRANSMISSION OF APPEAL BRIEF ORIGINALLY AND TIMELY FILED ON JULY 27, 2005

Honorable Commissioner:

This is a Retransmission of an Appeal Brief originally and timely filed on July 27, 2005, pursuant to 37 CFR § 41.37 in response to the Final Office Action of March 11, 2005 ("Office Action"), and pursuant to the Notice of Appeal filed May 27, 2005.

During a routine status check using Private PAIR, Applicants discovered that a timely filed Appeal Brief was never entered into the record in above-identified patent application. As evidenced by the enclosed copy of the postcard, the United States Patent and Trademark Office ('USPTO') received this Appeal Brief on July 29, 2005. Upon discovering that the USPTO did not enter the timely filed Appeal Brief, Applicants

immediately contacted the USPTO and have been advised to retransmit a copy of the original Appeal Brief. Applicants therefore retransmit herewith a copy of the original Appeal Brief along with copies of the five references that were also submitted with the original Appeal Brief and the return postcard demonstrating that the Appeal Brief was timely filed and received by the USPTO. Applicants respectfully request the acceptance and entry of the Appeal Brief.

The Commissioner is hereby authorized to charge or credit Deposit Account No. 09-0447 for any fees required or overpaid.

Respectfully submitted,

Date: August 10, 2006 By: ______________________

H. Artoush Ohanian
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ATTORNEY FOR APPLICANTS

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PATENT

The USPTO acknowledges with its official date stamp the filing of the documents indicated below, providing early notification of application no., as applicable

(10129)

Serial No. 10/015,280 Docket No. AUS920010823US1

Filing Date: 12/12/2001 Inventor(s): Michael W. Brown, et al.

Assignee: IBM

Title of Invention: Destination Device Based Callee Identification

☐ New Application: (Total Number of Pages: ______) Drawings: ______ Page(s)

☐ Declaration & Power of Attorney: ______ Page(s) ☐ Recordation & Assignment: ______ Page(s)

☐ Formal Drawings: ______ Page(s)

☐ Information Disclosure Statement ______ Page(s) ☐ Form PTO/SB/08A: ______ Page(s) ☐ Reference(s) ______

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☐ Notice of Appeal: ______ Page(s) ☒ Appeal Brief: 36 Page(s) ☒ Reference(s) 5

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TRANSMITTAL FORM *(to be used for all correspondence after initial filing)*		
	Application Number	10/015,280
	Filing Date	12/12/2001
	First Named Inventor	Michael W. Brown
	Art Unit	2645
	Examiner Name	Elahee, MD S
Total Number of Pages in This Submission: 36	Attorney Docket Number	AUS920010823US1

ENCLOSURES *(Check all that apply)*

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5 References (1 copy each); Postcard

Remarks

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Customer No. 34533.

SIGNATURE OF APPLICANT, ATTORNEY, OR AGENT

Firm or Individual name	H. Artoush Ohanian Reg. No. 46,022
Signature	[signature]
Date	July 27, 2005

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Typed or printed name	Catherine Berglund		
Signature	Catherine Berglund	Date	July 27, 2005

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of: Michael Wayne Brown, *et al.*	§	Group Art Unit: 2645
Serial No.: 10/015,280	§	Examiner: Elahee, MD S
Filed: December 12, 2001	§	Atty Docket No.: AUS920010823US1
Title: Destination Device Based Callee Identification	§	Customer No. 34533

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Date
Catherine Berglund
Catherine Berglund

APPEAL BRIEF

Honorable Commissioner:

This is an Appeal Brief filed pursuant to 37 CFR § 41.37 in response to the Final Office Action of March 11, 2005 ("Office Action"), and pursuant to the Notice of Appeal filed May 27, 2005.

REAL PARTY IN INTEREST

The real party in interest is the patent assignee, International Business Machines Corporation ("IBM"), a New York corporation having a place of business at Armonk, New York 10504.

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RELATED APPEALS AND INTERFERENCES

There are no related appeals or interferences.

STATUS OF CLAIMS

Claims 1-30 and 35-39 are pending in the case. All pending claims are on appeal.

STATUS OF AMENDMENTS

No amendments were submitted after final rejection. The claims as currently presented are included in the Appendix of Claims that accompanies this Appeal Brief.

SUMMARY OF CLAIMED SUBJECT MATTER

Applicants provide the following concise summary of the claimed subject matter according to 37 CFR§ 41.37(c)(1)(vii), including references to specification by page and line number and to the drawing(s) if any, by reference characters.

Methods, systems, and computer program products are provided for identifying a particular callee. Embodiments include detecting, at a destination device, a voice utterance of a callee; and identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call described for example at page 13, lines 6-16; page 24, lines 15-30; and Figure 4, S12-S22.

Methods, systems, and computer program products are provided for identifying a callee. Embodiments typically include detecting a biometric input at a biometric enabled destination device; identifying a callee identity associated with said biometric input at said destination device, such that said callee identity is transmittable as an authenticated

identity of said callee for a call described for example at page 19, line 27-page 20, line 26.

All such references to the specification identify descriptions and discussions that are part of the detailed descriptions of exemplary embodiments of the present invention in the present application. Such descriptions and discussions are not limitations of the claims in the present application. The only limitations of the claims are set forth in the claims themselves.

GROUNDS OF REJECTION

Claims 1, 4, 5, 7, 8, 10-12, 15, 16, 18, 19, 21-23, 26, 27, 29, 30, and 35-39 stand rejected under 35 U.S.C § 102(e) as being anticipated by Gallick (U.S. Patent No. 6,678,359). Claims 2, 13, and 24 stand rejected under 35 U.S.C. 103 (a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of Bartholomew et al. (U.S. Patent No. 6,167,119). Claim 3, 14, and 25 stand rejected under 35 U.S.C. 103(a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of McAllister (U.S. Patent No. 6,101,242). Claims 6, 17, and 28 stand rejected under 35 U.S.C. 103(a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of Timonen et al. (U.S. Pub. No. 2002/0058494). Claims 9 and 20 stand rejected under 35 U.S.C. 103(a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of Baker (U.S. Patent No. 5,533,109).

ARGUMENT

Claim Rejections – 35 U.S.C. § 102

Claims 1, 4, 5, 7, 8, 10-12, 15, 16, 18, 19, 21-23, 26, 27, 29, 30, and 35-39 stand rejected under 35 U.S.C § 102(e) as being anticipated by Gallick (U.S. Patent No. 6,678,359). To anticipate under 35 U.S.C. § 102(e), two basic requirements must be met. The first requirement of anticipation is that Gallick must disclose each and every element as set

forth in Applicants' claims. The second requirement of anticipation is that Gallick must enable Applicants' claims. Gallick does not meet either requirement and therefore does not anticipate Applicants' claims.

<u>Gallick Does Not Disclose Each and Every Element of Applicants' Claims</u>

"A claim is anticipated only if each and every element as set forth in the claim is found, either expressly or inherently described, in a single prior art reference." *Verdegaal Bros. v. Union Oil Co. of California*, 814 F.2d 628, 631, 2 USPQ2d 1051, 1053 (Fed. Cir. 1987). Independent claim 1 claims:

> A method for identifying a particular callee, said method comprising:
>
> detecting, at a destination device, a voice utterance of a callee; and
>
> identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call.

The Office Action states that Gallick discloses "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call" at column 3, lines 64-67; column 6, lines 3-14, 51-56; and column 6, lines 51-53. Column 3, lines 64-67 actually states: "If, however, the individual answering the call at the called facility has been identified that identity will be transmitted back to the calling subscriber." Column 6, lines 3-14, discloses using voice identification software to analyze the speech of an answering party at the called facility. Column 6, lines 51-56, discloses capturing utterances of the called party and sending the captured utterances to caller verification routines for an attempted identification of the called party.

The cited section of Gallick does not disclose "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call" as claimed in the present application. Instead Gallick discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Therefore, Gallick does not disclose each and every element of independent claim 1 and does not anticipate claim 1. Independent claim 1 is patentable and should be allowed. Applicants therefore request reversal of the rejection of claim 1.

Dependent claims 4, 5, 7, 8, 10 and 11 depend from independent claim 1 and include all of the limitations of claim 1. Because Gallick does not disclose each and every element of claim 1, Gallick does not disclose each and every element of claims 4, 5, 7, 8, 10 and 11. Gallick therefore does not anticipate claims 4, 5, 7, 8, 10 and 11. Claims 4, 5, 7, 8, 10 and 11 are also patentable and should be allowed. Applicants therefore request reversal of the rejection of claims 4, 5, 7, 8, 10 and 11.

<u>Gallick Does Not Disclose Each and Every Element of Independent Claims 12 and 23</u>

Independent claims 12 and 23 claim a system and computer program product corresponding to method claim 1. More particularly, independent claims 12 and 23 claim system and computer program products for identifying a particular callee. The Office Action rejects claims 12 and 23 on the same grounds as claim 1. In response, Applicants respectfully note that for the same reasons that Gallick does not disclose each and every element of claim 1, Gallick does not disclose each and every element of claims 12 and 23. Gallick therefore does not anticipate system and computer program products claims 12 and 23, respectively, and claims 12 and 23 are also patentable and should be allowed. Applicants therefore request reversal of the rejection of claims 12 and 23.

Dependent claims 15, 16, 18, 19, 21, 22, 26, 27, 29, and 30 depend from independent claims 12 and 23 and include all of the limitations of claims 12 and 23. Because Gallick

does not disclose each and every element of claims 12 and 23, Gallick also does not disclose each and every element of dependent claims 15, 16, 18, 19, 21, 22, 26, 27, 29 and 30. Gallick therefore does not anticipate claims 15, 16, 18, 19, 21, 22, 26, 27, 29 and 30 and claims 15, 16, 18, 19, 21, 22, 26, 27, 29 and 30 are patentable and should be allowed. Applicants request reversal of the rejection of claims 15, 16, 18, 19, 21, 22, 26, 27, 29 and 30.

Gallick Does Not Disclose Each And Every Element of Independent Claim 35

Gallick does not disclose each and every element of claim 35. Independent claim 35 claims:

> A method for identifying a callee, said method comprising:
>
> detecting a biometric input at a biometric enabled destination device;
>
> identifying a callee identity associated with said biometric input at said destination device, such that said callee identity is transmittable as an authenticated identity of said callee for a call.

The Office Action states that Gallick discloses detecting biometric input at a biometric enabled destination device at Figure 1, Figure 2b, column 1, line 50-column 2, line 11, column 3, lines 64-67 and col. 6, lines 3-14, 51-56. Figure 1 of Gallick actually sets forth a block diagram of a communications system. Figure 2b actually sets forth a flow chart illustrating the method described in part at column 6, lines 3-14 and 51-56. Column 1, line 50-column 2, line 11 actually describes called party identification particularly adapted to VoIP calls. Column 3, lines 64-67 actually states: "If, however, the individual answering the call at the called facility has been identified that identity will be transmitted back to the calling subscriber." Column 6, lines 3-14 discloses using voice identification software to analyze the speech of an answering party at the called facility. Column 6, lines 51-56 discloses capturing utterances of the called party and sending the

captured utterances to caller verification routines for an attempted identification of the called party.

The cited sections of Gallick do not disclose "identifying a callee identity associated with said biometric input at said destination device, such that said callee identity is transmittable as an authenticated identity of said callee for a call." Instead, Gallick discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Therefore, Gallick does not disclose each and every element of independent claim 35 and does not anticipate claim 35. Independent claim 35 is patentable and should be allowed. Applicants request reversal of the rejection of claim 35.

Dependent claim 36 depends from independent claim 35 and includes all of the limitations of independent claim 35. Because Gallick does not disclose each and every element of independent claim 35, Gallick does not disclose each and every element of dependent claim 36 and does not anticipate claim 36. Claim 36 is therefore also patentable and should be allowed. Applicants request reversal of the rejection of claim 36.

<u>Gallick Does Not Disclose Each and Every Element of Independent Claims 37 and 39</u>

Independent claims 37 and 39 claim system and computer program product claims corresponding to method claim 35. More particularly, independent claims 37 and 39 recite system and computer program product for identifying a callee. Claims 37 and 39 stand rejected for the same reason as independent claim 35. Because Gallick does not disclose each and every element of method claim 35, Gallick does not disclose each and every element of claims 37 and 39 and therefore does not anticipate claims 37 and 39. Claim 37 and 39 are therefore also patentable and should be allowed. Applicants request reversal of the rejection of claims 37 and 39.

Dependent claim 38 depends from independent claim 37 and includes all of the limitations of independent claim 37. Because Gallick does not disclose each and every element of independent claim 37, Gallick does not disclose each and every element of dependent claim 38 and does not anticipate claim 38. Claim 38 is therefore also patentable and should be allowed. Applicants request reversal of the rejection of claim 38.

<u>Gallick Is Not An Enabling</u>
<u>Disclosure of Applicants' Claims</u>

There are two required aspects of anticipation. Not only must Gallick disclose each and every element of the claims of the present invention within the meaning of *Verdegaal* in order to anticipate the claims, but Gallick must also be an enabling disclosure of the claims of the present invention within the meaning of *In re Hoeksema*. The Appellants' claims in *Hoeksema* were rejected because an earlier patent disclosed a close structural similarity to appellant's chemical compound. The court in Hoeksema stated: "We think it is sound law, consistent with the public policy underlying our patent law, that before any publication can amount to a statutory bar to the grant of a patent, its disclosure must be such that a skilled artisan could take its teachings in combination with his own knowledge of the particular art and be in possession of the invention." The meaning of *Hoeksema* for the present case is that to anticipate under 35 USC 102(e) Gallick must place one of skill in the art in possession of Applicants' claims.

Gallick does not enable independent claim 1. Independent claim 1 claims:

> A method for identifying a particular callee, said method comprising:
>
> detecting, at a destination device, a voice utterance of a callee; and

identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call.

The Office Action states that Gallick discloses "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call" at column 3, lines 64-67; column 6, lines 3-14, 51-56; and column 6, lines 51-53. Column 3, lines 64-67, actually states: "If, however, the individual answering the call at the called facility has been identified that identity will be transmitted back to the calling subscriber." Column 6, lines 3-14, discloses using voice identification software to analyze the speech of an answering party at the called facility. Column 6, lines 51-56, discloses capturing utterances of the called party and sending the captured utterances to a caller verification routines for an attempted identification of the called party.

The cited sections of Gallick do not place one of skill in the art in possession of "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call" as claimed in the present application. Instead Gallick discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Therefore, Gallick does not place one of skill in the art in possession of independent claim 1 and does not anticipate claim 1. Independent claim 1 is patentable and should be allowed. Applicants request reversal of the rejection of claim 1.

Dependent claims 4, 5, 7, 8, 10 and 11 depend from independent claim 1 and include all of the limitations of claim 1. Because Gallick does not place one of skill in the art in possession of claim 1, Gallick does not place one of skill in the art in possession of claims 4, 5, 7, 8, 10 and 11. Gallick therefore does not anticipate claims 4, 5, 7, 8, 10 and 11. Claims 4, 5, 7, 8, 10 and 11 are also patentable and should be allowed. Applicants request reversal of the rejection of claims 4, 5, 7, 8, 10 and 11.

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Gallick Does Not Enable
Independent Claims 12 and 23

Independent claims 12 and 23 claim a system and computer program product corresponding to method claim 1. More particularly, independent claims 11and 23 claim system and computer program products for identifying a particular callee. The Office Action rejects claims 12 and 23 on the same grounds as claim 1. In response, Applicants respectfully note that for the same reasons Gallick does not place one of skill in the art in possession of claim 1, Gallick does not place one of skill in the art in possession of claims 12 and 23. Gallick therefore does not anticipate system and computer program products claims 12 and 23, respectively, and claims 12 and 23 are also patentable and should be allowed. Applicants request reversal of the rejection of claims 12 and 23.

Dependent claims 15, 16, 18, 19, 21, 22, 26, 27, 29 and 30 depend from independent claims 12 and 23 and include all of the limitations of claims 12 and 23. Because Gallick does not place on of skill in the art in possession of claims 12 and 23, Gallick does not place one of skill in the art in possession of claims 15, 16, 18, 19, 21, 22, 26, 27, 29 and 30. Gallick therefore does not anticipate claims 15, 16, 18, 19, 21, 22, 26, 27, 29 and 30. Claims 15, 16, 18, 19, 21, 22, 26, 27, 29 and 30 are also patentable and should be allowed. Applicants request reversal of the rejection of claims 15, 16, 18, 19, 21, 22, 26, 27, 29 and 30.

Gallick Does Not Enable Independent Claim 35

Gallick is not an enabling disclosure of independent claim 35. Independent claim 35 claims:

> A method for identifying a callee, said method comprising:
>
> detecting a biometric input at a biometric enabled destination device;

identifying a callee identity associated with said biometric input at said destination device, such that said callee identity is transmittable as an authenticated identity of said callee for a call.

The Office Action states that Gallick discloses detecting biometric input at a biometric enabled destination device at Figure 1, Figure 2b, column 1, line 50-column 2, line 11, column 3, lines 64-67, and col. 6, lines 3-14, 51-56. Figure 1 of Gallick actually sets forth a block diagram of a communications system. Figure 2b actually sets forth a flow chart illustrating the method described in part at column 6, lines 3-14, and 51-56. Column 1, line 50-column 2, line 11, actually describes called party identification particularly adapted to VoIP calls. Column 3, lines 64-67, actually states: "If, however, the individual answering the call at the called facility has been identified that identity will be transmitted back to the calling subscriber." Column 6, lines 3-14, discloses using voice identification software to analyze the speech of an answering party at the called facility. Column 6, lines 51-56, discloses capturing utterances of the called party and sending the captured utterances to caller verification routines for an attempted identification of the called party.

The cited sections of Gallick do not place one of skill in the art in possession of "identifying a callee identity associated with said biometric input at said destination device, such that said callee identity is transmittable as an authenticated identity of said callee for a call." Instead, Gallick discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Therefore, Gallick does not place one of skill in the art in possession of independent claim 35 and does not anticipate claim 35. Independent claim 35 is patentable and should be allowed. Applicants request reversal of the rejection of claim 35.

Dependent claim 36 depends from independent claim 35 and includes all of the limitations of independent claim 35. Because Gallick does not place one of skill in the

art in possession of independent claim 35, Gallick does not disclose each and every element of dependent claim 36 and does not anticipate claim 36. Claim 36 is therefore also patentable and should be allowed. Applicants request reversal of the rejection of claims 36.

<u>Gallick Does Not Enable Independent Claims 37 and 39</u>

Independent claims 37 and 39 claim system and computer program product claims corresponding to method claim 35. More particularly, independent claims 37 and 39 recite system and computer program product for identifying a callee. Claims 37 and 39 stand rejected for the same reason as independent claim 35. Because Gallick does not place one of skill in the art in possession of claim 35, Gallick does not place one of skill in the art in possession of claims 37 and 39 and therefore does not anticipate claims 37 and 39. Claim 37 and 39 are therefore also patentable and should be allowed. Applicants request reversal of the rejection of claims 37 and 39.

Dependent claim 38 depends from independent claim 37 and includes all of the limitations of independent claim 37. Because Gallick does not place one of skill in the art in possession of independent claim 37, Gallick does not place one of skill in the art in possession of dependent claim 38 and does not anticipate claim 38. Claim 38 is therefore also patentable and should be allowed. Applicants request reversal of the rejection of claim 38.

Claim Rejections – 35 U.S.C. § 103

Claims 2, 13, and 24 stand rejected under 35 U.S.C. 103(a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of Bartholomew et al. (U.S. Patent No. 6,167,119). Claim 3, 14, and 25 stand rejected under 35 U.S.C. 103(a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of McAllister (U.S. Patent No. 6,101,242). Claims 6, 17, and 28 stand rejected under 35 U.S.C. 103(a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of Timonen et al. (U.S.

Pub. No. 2002/0058494). Claims 9 and 20 stand rejected under 35 U.S.C. 103(a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of Baker (U.S. Patent No. 5,533,109). Applicants respectfully traverse each rejection. Not one of the proposed combinations can establish a prima facie case of obviousness.

To establish a prima facie case of obviousness, three basic criteria must be met. *Manual of Patent Examining Procedure* §2142. The first element of a prima facie case of obviousness under 35 U.S.C. § 103 is that there must be a suggestion or motivation to combine Gallick and Bartholomew, McAllister, Timonen, or Baker. *In re Vaeck*, 947 F.2d 488, 493, 20 USPQ2d 1438, 1442 (Fed. Cir. 1991). The second element of a prima facie case of obviousness under 35 U.S.C. § 103 is that there must be a reasonable expectation of success in the proposed combinations of Gallick and Bartholomew, McAllister, Timonen, or Baker. *In re Merck & Co., Inc.*, 800 F.2d 1091, 1097, 231 USPQ 375, 379 (Fed. Cir. 1986). The third element of a prima facie case of obviousness under 35 U.S.C. § 103 is that the proposed combinations of Gallick and Bartholomew, McAllister, Timonen, or Baker must teach or suggest all of Applicants' claim limitations. *In re Royka*, 490 F.2d 981, 985, 180 USPQ 580, 583 (CCPA 1974). The proposed combinations of Gallick and Bartholomew, McAllister, Timonen, or Baker do not establish even a prima facie case of obviousness and therefore cannot support a rejection under 35 U.S.C. 103. The rejections should therefore be reversed and the case allowed.

Gallick and Bartholomew

Claims 2, 13, and 24 stand rejected under 35 U.S.C § 103(a) as unpatentable over Gallick in view of Bartholomew. The combination of Gallick and Bartholomew cannot establish a prima facie case of obviousness because the proposed combination does not teach each and every element of claims 2, 13, and 24, there is no suggestion or motivation to make the proposed combination, and there is no reasonable expectation of success in the proposed combination. Applicants request reversal of the rejection of claims 2, 13, and 24.

No Suggestion or Motivation to
Combine of Gallick and Bartholomew

To establish a prima facie case of obviousness, there must be a suggestion or motivation to combine Gallick and Bartholomew. *In re Vaeck*, 947 F.2d 488, 493, 20 USPQ2d 1438, 1442 (Fed. Cir. 1991). There is no suggestion or motivation to combine Gallick and Bartholomew, because Bartholomew teaches away from Applicants' claims. Teaching away from the claims in the present application is a *per se* demonstration of lack of prima facie obviousness. *In re Dow Chemical Co.*, 837 F.2d 469, 5 U.S.P.Q.2d 1529 (Fed. Cir. 1988); *In re Fine*, 837 F.2d 1071, 5 U.S.P.Q.2d 1596 (Fed. Cir. 1988); *In re Neilson*, 816 F.2d 1567, 2 U.S.P.Q.2d 1525 (Fed. Cir. 1987). Independent claim 1 claims a method for identifying a particular callee that includes "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call." Bartholomew discloses using an intermediary intelligent peripheral ('IP')—not a destination device and therefore, teaches away from Applicants' claims. *See for example*, Bartholomew, column 11, line 62 – column 12, line 50. As such, the proposed combination of Gallick and Bartholomew cannot establish a prima facie case of obviousness. The rejection should be reversed and the case allowed.

No Reasonable Expectation of Success
In the Proposed Combination of Gallick and Bartholomew

To establish a prima facie case of obviousness, there must be a reasonable expectation of success in the proposed combination of Gallick and Bartholomew. *In re Merck & Co., Inc.*, 800 F.2d 1091, 1097, 231 USPQ 375, 379 (Fed. Cir. 1986). There is no reasonable expectation of success to combine Gallick and Bartholomew because the proposed combination changes the principle of operation of Bartholomew. "If the proposed modification or combination of the prior art would change the principle of operation of the prior art invention being modified, then the teachings of the references are not sufficient to render the claims prima facie obvious." *In re Ratti*, 270 F.2d 810, 123

USPQ 349 (CCPA 1959). Bartholomew discloses an intermediary intelligent peripheral –not a destination device. *See for example,* Bartholomew, column 11, line 62 – column 12, line 50. To modify Bartholomew to teach "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call," would change the stated principle operation of Bartholomew. The proposed combination and Gallick and Bartholomew therefore cannot establish a prima facie case of obviousness. The rejection should be reversed and the case allowed.

<u>The Combination of Gallick and Bartholomew
Does Not Teach All of Applicants' Claim Limitations</u>

To establish a prima facie case of obviousness, the proposed combination of Gallick and Bartholomew must teach or suggest all of Applicants' claim limitations. *In re Royka*, 490 F.2d 981, 985, 180 USPQ 580, 583 (CCPA 1974). Claims 2, 13, and 24 include identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call. As demonstrated above, Gallick does not teach this limitation. Gallick instead discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Bartholomew also does not teach this limitation. Bartholomew instead discloses an intermediary intelligent peripheral –not a destination device. *See for example,* Bartholomew, column 11, line 62 – column 12, line 50. As such, the combination of Gallick and Bartholomew cannot establish a prima facie case and the rejection should be reversed.

<u>Gallick and McAllister</u>

Claim 3, 14, and 25 stand rejected under 35 U.S.C. 103(a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of McAllister (U.S. Patent No. 6,101,242). The combination of Gallick and McAllister cannot establish a prima facie case of

obviousness because the proposed combination does not teach each and every element of claims 3, 14, and 25, there is no suggestion or motivation to make the proposed combination, and there is no reasonable expectation of success in the proposed combination. Applicants request reversal of the rejection of claims 3, 14, and 25.

No Suggestion or Motivation to
Combine of Gallick and McAllister

To establish a prima facie case of obviousness, there must be a suggestion or motivation to combine Gallick and McAllister. *In re Vaeck*, 947 F.2d 488, 493, 20 USPQ2d 1438, 1442 (Fed. Cir. 1991). There is no suggestion or motivation to combine Gallick and McAllister, because McAllister teaches away from Applicants' claims. Teaching away from the claims in the present application is a *per se* demonstration of lack of prima facie obviousness. *In re Dow Chemical Co.*, 837 F.2d 469, 5 U.S.P.Q.2d 1529 (Fed. Cir. 1988); *In re Fine*, 837 F.2d 1071, 5 U.S.P.Q.2d 1596 (Fed. Cir. 1988); *In re Neilson*, 816 F.2d 1567, 2 U.S.P.Q.2d 1525 (Fed. Cir. 1987). Independent claim 1 claims method for identifying a particular callee that includes "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call." McAllister discloses an intermediary intelligent peripheral—not a destination device and therefore teaches away from Applicants' claims. *See for example*, McAllister, column 12, line 47 – column 13, line 35. As such, the proposed combination of Gallick and McAllister cannot establish a prima facie case of obviousness. The rejection should be reversed and the case allowed.

No Reasonable Expectation of Success
In the Proposed Combination of Gallick and McAllister

To establish a prima facie case of obviousness, there must be a reasonable expectation of success in the proposed combination of Gallick and McAllister. *In re Merck & Co., Inc.*, 800 F.2d 1091, 1097, 231 USPQ 375, 379 (Fed. Cir. 1986). There is no reasonable

expectation of success to combine Gallick and McAllister because the proposed combination changes the principle of operation of McAllister. "If the proposed modification or combination of the prior art would change the principle of operation of the prior art invention being modified, then the teachings of the references are not sufficient to render the claims prima facie obvious." *In re Ratti,* 270 F.2d 810, 123 USPQ 349 (CCPA 1959). McAllister discloses an intermediary intelligent peripheral. *See for example,* McAllister, column 11, line 62 – column 12, line 50. To modify McAllister to teach "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call," would change the principle operation of McAllister. The proposed combination and Gallick and McAllister therefore cannot establish a prima facie case of obviousness. The rejection should be reversed and the case allowed.

The Combination of Gallick and McAllister Does Not Teach All of Applicants' Claim Limitations

To establish a prima facie case of obviousness, the proposed combination of Gallick and McAllister must teach or suggest all of Applicants' claim limitations. *In re Royka,* 490 F.2d 981, 985, 180 USPQ 580, 583 (CCPA 1974). Claims 2, 13, and 24 include "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call." As demonstrated above, Gallick does not teach this limitation. Gallick instead discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. McAllister also does not disclose this limitation. McAllister instead discloses an intermediary intelligent peripheral—not a destination device and therefore teaches away from Applicants' claims. *See for example,* McAllister, column 12, line 47 – column 13, line 35. As such, the combination of Gallick and McAllister cannot establish a prima facie case and the rejection should be reversed.

Gallick and Timonen

Claims 6, 17, and 28 stand rejected under 35 U.S.C. 103(a) as being unpatentable over Gallick (U.S. Patent No. 6,678,359) in view of Timonen et al. (U.S. Pub. No. 2002/0058494). The combination of Gallick and Timonen cannot establish a prima facie case of obviousness because the proposed combination does not teach each and every element of claims 6, 17, and 28, there is no suggestion or motivation to make the proposed combination, and there is no reasonable expectation of success in the proposed combination. Applicants request reversal of the rejection of claims 6, 17, and 28.

The Combination of Gallick and Timonen
Does Not Teach All of Applicants' Claim Limitations

To establish a prima facie case of obviousness, the proposed combination of Gallick and Timonen must teach or suggest all of Applicants' claim limitations. *In re Royka*, 490 F.2d 981, 985, 180 USPQ 580, 583 (CCPA 1974). Rejected claims 6, 17, and 28 depend from independent claims 1, 12, and 23 and include the limitations "identifying, at said destination device, a callee identity associated with said utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call." As demonstrated above, Gallick does not teach this limitation. Gallick instead discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Timonen does not teach what Gallick lacks. Instead, Timonen at Figure 3 and page 6, paragraphs 0055 and 0056, discloses an encrypted message containing a digital signature sent to a third party. Such an encrypted message is not identifying, at said destination device, a callee identity associated with said utterance. In fact, a destination device is not even mentioned in the cited sections of Timonen. Because the combination of Gallick and Timonen does not teach each and every limitation of claims 6, 17, and 28, the combination of Gallick and Timonen cannot establish a prima facie case of obviousness. The rejection should be reversed and the case allowed.

No Suggestion or Motivation to Modify Gallick or Combine of Gallick and Timonen

To establish a prima facie case of obviousness, there must be a suggestion or motivation to modify Gallick. *In re Vaeck*, 947 F.2d 488, 493, 20 USPQ2d 1438, 1442 (Fed. Cir. 1991). The suggestion or motivation to modify Gallick must come from the teaching of Gallick itself and the Examiner must explicitly point to the teaching within Gallick suggesting the proposed modification. Absent such a showing, the Examiner has impermissibly used "hindsight" occasioned by Applicants' own teaching to reject the claims. *In re Surko,* 11 F.3d 887, 42 U.S.P.Q.2d 1476 (Fed. Cir. 1997); *In re Vaeck*, 947 F.2d 488m 20 U.S.P.Q.2d 1438 (Fed. Cir. 1991); *In re Gorman*, 933 F.2d 982, 986, 18 U.S.P.Q.2d 1885, 1888 (Fed. Cir. 1991); *In re Bond*, 910 F.2d 831, 15 U.S.P.Q.2d 1566 (Fed. Cir. 1990); *In re Laskowski,* 871 F.2d 115, 117, 10 U.S.P.Q.2d 1397, 1398 (Fed. Cir. 1989). The Office Action fails to point to teaching within either Gallick or Timonen suggesting their combination. Without more, the rejection should be reversed.

There is in fact no teaching in either Gallick or Timonen suggesting the proposed combination. As discussed above, Gallick discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Timonen at Figure 3 and page 6, paragraphs 0055, 0056 discloses an encrypted message containing a digital signature sent to a third party. There is no suggestion to combine the encrypted message of Timonen with the voice identification recognizer of Gallick. The combination of Gallick and Timonen therefore cannot support a prima facie case of obviousness. The rejection should be reversed and the case allowed.

No Reasonable Expectation of Success In the Proposed Combination of Gallick and Timonen

To establish a prima facie case of obviousness, there must be a reasonable expectation of success in the proposed combination of Gallick and Timonen. *In re Merck & Co., Inc.,*

800 F.2d 1091, 1097, 231 USPQ 375, 379 (Fed. Cir. 1986). There is no reasonable expectation of success to combine Gallick and Timonen. As discussed above, Gallick discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Timonen at Figure 3 and page 6, paragraphs 0055, 0056 discloses an encrypted message containing a digital signature sent to a third party. The encrypted message of Timonen combined with the voice identification recognizer of Gallick will not work to identify, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call as claimed in the present application. As such, the proposed combination and Gallick and Timonen therefore cannot establish a prima facie case of obviousness. The rejection should be reversed and the case allowed.

<u>Gallick and Baker</u>

Claims 9 and 20 stand rejected under 35 U.S.C. § 103 as unpatentable over Gallick in view of Baker. The combination of Gallick and Baker also cannot establish a prima facie case of obviousness because the proposed combination does not teach each and every element of claims 9 and 20, there is no suggestion or motivation to make the proposed combination, and there is no reasonable expectation of the success in the proposed combination. Applicants request reversal of the rejection of claims 9 and 20.

<u>The Combination of Gallick and Baker</u>
<u>Does Not Teach All of Applicants' Claim Limitations</u>

To establish a prima facie case of obviousness, the proposed combination of Gallick and Baker must teach or suggest all of Applicants' claim limitations. *In re Royka*, 490 F.2d 981, 985, 180 USPQ 580, 583 (CCPA 1974). The proposed combination of Gallick and Baker does not teach each and every element of claims 9 and 20. Claim 9 depends from independent claim 1 and includes the limitation "identifying, at said destination device, a callee identity associated with said utterance, such that said callee identity is

transmittable as an authenticated identity of said callee for a call." Claim 20 depends from claim 12 and includes the limitation "means for identifying a callee identity associated with said voice utterance at said destination device, such that said callee identity is transmittable as an authenticated identity of said callee for a call." As discussed above, Gallick does not disclose "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call" as claimed in the present application. Gallick instead discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Baker does not teach what Gallick lacks. Baker instead discloses a telecommunications system with a PBX. *See for example* Baker, abstract, column 2 lines 40-50. In fact, Baker also does not even address authentication. Because the proposed combination of Gallick and Baker fails to teach every element of claims 9 and 20, the combination cannot establish a prima face case of obviousness. The rejection should be reversed and the case allowed.

<u>No Suggestion or Motivation to Modify Gallick or</u>
<u>Combine of Gallick and Baker</u>

To establish a prima facie case of obviousness, there must be a suggestion or motivation to modify Gallick. *In re Vaeck*, 947 F.2d 488, 493, 20 USPQ2d 1438, 1442 (Fed. Cir. 1991). The suggestion or motivation to modify Gallick must come from the teaching of Gallick itself and the Examiner must explicitly point to the teaching within Gallick suggesting the proposed modification. Absent such a showing, the Examiner has impermissibly used "hindsight" occasioned by Applicants' own teaching to reject the claims. *In re Surko*, 11 F.3d 887, 42 U.S.P.Q.2d 1476 (Fed. Cir. 1997); *In re Vaeck*, 947 F.2d 488m 20 U.S.P.Q.2d 1438 (Fed. Cir. 1991); *In re Gorman*, 933 F.2d 982, 986, 18 U.S.P.Q.2d 1885, 1888 (Fed. Cir. 1991); *In re Bond*, 910 F.2d 831, 15 U.S.P.Q.2d 1566 (Fed. Cir. 1990); *In re Laskowski*, 871 F.2d 115, 117, 10 U.S.P.Q.2d 1397, 1398 (Fed. Cir. 1989). The Office Action fails to point to teaching within either Gallick or Baker suggesting their combination. Without more, the rejection should be reversed.

There is in fact no teaching in either Gallick or Baker suggesting "identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call." As discussed above, Gallick discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Baker discloses a telecommunications system with a PBX. *See for example* Baker, abstract, column 2 lines 40-50. There is no suggestion in either Baker or Gallick to combine the telecommunications system of Baker with the voice identification recognizer of Gallick. The combination of Gallick and Baker therefore cannot support a prima facie case of obviousness. The rejection should be reversed and the case allowed.

<u>No Reasonable Expectation of Success</u>
<u>In the Proposed Combination of Gallick and Baker</u>

To establish a prima facie case of obviousness, there must be a reasonable expectation of success in the proposed combination of Gallick and Baker. *In re Merck & Co., Inc.*, 800 F.2d 1091, 1097, 231 USPQ 375, 379 (Fed. Cir. 1986). There is no reasonable expectation of success to combine Gallick and Baker. As discussed above, Gallick discloses a voice identification recognizer located on a personal computer or on a server on the network where the softphone resides. Gallick, column 6, lines 4-20. Baker discloses a telecommunications system with a PBX. *See for example* Baker, abstract, column 2 lines 40-50. The telecommunications system of Baker combined with the voice identification recognizer of Gallick will not work to identify, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call as claimed in the present application. As such, the proposed combination and Gallick and Baker therefore cannot establish a prima facie case of obviousness. The rejection should be reversed and the case allowed.

The Four Factual Inquires Required By The Supreme Court For An Obviousness Rejection Have Not Been Properly Considered, Determined, and Applied

Establishing a prima facie case of obviousness for claims 2, 3, 6, 9, 13, 14, 17, 20, 24, 25, and 28, which has not been accomplished, is not the end of obviousness analysis, it is the beginning. The rejection of these Applicants' claims under 35 U.S.C. § 103 are deficient because the proper factual inquiries have not been considered, determined, and applied as required by the Supreme Court in *Graham v. John Deere*. The rejection should therefore be reversed and the case allowed.

The Manual of Patent Examining Procedure §2141 explicitly states:

> Patent examiners carry the responsibility of making sure that the standard of patentability enunciated by the Supreme Court and by the Congress is applied in each and every case. The Supreme Court in Graham v. John Deere, 383 U.S. 1, 148 USPQ 459 (1966), stated:
>
>> Under Section 103, the scope and content of the prior art are to be determined; differences between the prior art and the claims at issue are to be ascertained; and the level of ordinary skill in the pertinent art resolved. Against this background, the obviousness or nonobviousness of the subject matter is determined. Such secondary considerations as commercial success, long felt but unsolved needs, failure of others, etc., might be utilized to give light to the circumstances surrounding the origin of the subject matter sought to be patented. As indicia of obviousness or nonobviousness, these inquiries may have relevancy. . .

> This is not to say, however, that there will not be difficulties in applying the nonobviousness test. What is obvious is not a question upon which there is likely to be uniformity of thought in every given factual context. The difficulties, however, are comparable to those encountered daily by the courts in such frames of reference as negligence and scienter, and should be amenable to a case-by-case development. We believe that strict observance of the requirements laid down here will result in that uniformity and definitiveness which Congress called for in the 1952 Act.

Office policy has consistently been to follow Graham v. John Deere Co. in the consideration and determination of obviousness under 35 U.S.C. 103. As quoted above, the four factual inquires enunciated therein as a background for determining obviousness are briefly as follows:

(A) Determining of the scope and contents of the prior art;

(B) Ascertaining the differences between the prior art and the claims in issue;

(C) Resolving the level of ordinary skill in the pertinent art; and

(D) Evaluating evidence of secondary considerations.

Manual of Patent Examining Procedure §2141.

In over three years of prosecution, the Examiner has yet to even mention the four factual inquiries required by the Supreme Court in *Graham v. John Deere,* and all four factual

inquires have not been properly considered, determined, and applied in any of the office actions in this case.

The first factual inquiry that has not been properly considered and determined is ascertaining the differences between the prior art and the claims in issue. More particularly, in the present office action the Examiner has only identified elements in Applicants' claims not found in Gallick and then attempted to find a similar element in Bartholomew, McAllister, Timonen, or Baker to support an obviousness rejection. Such analysis is improper and incomplete. "Ascertaining the differences between the prior art and the claims at issue requires interpreting the claim language, and considering both the invention and the prior art references as a whole." MPEP §2141.02. Furthermore, "[i]n determining the differences between the prior art and the claims, the question under 35 U.S.C. 103 is not whether the differences <u>themselves</u> would have been obvious, but whether the claimed invention <u>as a whole</u> would have been obvious." *Id.*, citing *Stratoflex, Inc. v. Aeroquip Corp.*, 713 F.2d 1530 (Fed. Cir. 1983). The analysis of the present office action is improper and incomplete because Examiner has not determined whether Applicants claims as a whole would have been obvious in view of a combination of Gallick and Bartholomew, McAllister, Timonen, or Baker as required by the Manual of Patent Examining Procedure. In fact, the Examiner has not even mentioned how the claim as a whole would be obvious in rejecting any claim. As such, the obviousness rejections should be reversed and the case should be allowed.

The second factual inquiry that has not been properly considered, determined, and applied is resolving the level of ordinary skill in the pertinent art. "The importance of resolving the level of ordinary skill in the art lies in the necessity of maintaining objectivity in the obviousness inquiry." MPEP §2141.03 citing *Ryko Mfg. Co. v. Nu-Star, Inc.*, 950 F.2d 714, 718, 21 USPQ2d 1053, 1057 (Fed. Cir. 1991). "The examiner must ascertain what would have been obvious to one of ordinary skill in the art at the time the invention was made, and not to the inventor, a judge, a layman, those skilled in remote arts, or to geniuses in the art at hand." *Id.* citing *Environmental Designs, Ltd. v. Union Oil Co.*, 713 F.2d 693, 218 USPQ 865 (Fed. Cir. 1983), cert. denied, 464 U.S. 1043 (1984).

"Factors that may be considered in determining level of ordinary skill in the art include (1) the educational level of the inventor; (2) type of problems encountered in the art; (3) prior art solutions to those problems; (4) rapidity with which innovations are made; (5) sophistication of the technology; and (6) educational level of active workers in the field." *Id.* citing *Environmental Designs, Ltd. v. Union Oil Co.*, 713 F.2d 693, 696, 218 USPQ 865, 868 (Fed. Cir. 1983), cert. denied, 464 U.S. 1043 (1984). The present office action fails to apply a single factor to consider in determining the level of ordinary skill in the art. In fact, in over three years of prosecution and five office actions, no analysis at all considering the level of one of ordinary skill in the art for the instant case has been performed. The rejection is therefore deficient and the rejection should be reversed.

Conclusion

To anticipate claims 1, 4, 5, 7, 8, 10-12, 15, 16, 18, 19, 21-23, 26, 27, 29, 30, and 35-39, Gallick must disclose each and every element as set forth in the claims 1, 4, 5, 7, 8, 10-12, 15, 16, 18, 19, 21-23, 26, 27, 29, 30, and 35-39 and be an enabling disclosure of claims 1, 4, 5, 7, 8, 10-12, 15, 16, 18, 19, 21-23, 26, 27, 29, 30, and 35-39. Because Gallick does not disclose or place one of ordinary skill in the art in possession Applicants' claims, Gallick cannot anticipate claims 1, 4, 5, 7, 8, 10-12, 15, 16, 18, 19, 21-23, 26, 27, 29, 30, and 35-39 within the meaning of 35 USC § 102. Gallick alone or in combination with Bartholomew, McAllister, Timonen, or Baker does not establish a prima facie case of obviousness according to 35 USC § 103. The proposed combinations of Gallick and Bartholomew, McAllister, Timonen, or Baker fail to establish a prima facie case of obviousness because the proposed combinations present no suggestions or motivation to combine the references, there is no reasonable expectation of success in the proposed combinations, and the proposed combinations do not teach all of Applicant's claim limitations. Applicants therefore respectfully request the allowance of claims 1-30 and 35-39.

In view of the forgoing arguments, reversal on all grounds of rejection is requested.

The Commissioner is hereby authorized to charge or credit Deposit Account No. 09-0447 for any fees required or overpaid.

Respectfully submitted,

Date: July 27, 2005 By: ______________________

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APPENDIX OF CLAIMS
ON APPEAL IN PATENT APPLICATION OF
MICHAEL WAYNE BROWN, *ET AL.*, SERIAL NO. 10/015,280

CLAIMS

What is claimed is:

1. A method for identifying a particular callee, said method comprising:

   detecting, at a destination device, a voice utterance of a callee; and

   identifying, at said destination device, a callee identity associated with said voice utterance, such that said callee identity is transmittable as an authenticated identity of said callee for a call.

2. The method for identifying a particular callee according to claim 1, further comprising:

   prompting said callee, from said destination device, to provide said voice utterance.

3. The method for identifying a particular callee according to claim 1, further comprising:

   prompting said callee to enter an additional input to verify said callee identity.

4. The method for identifying a particular callee according to claim 1, wherein identifying a callee identity further comprises:

   extracting speech characteristics from said voice utterance; and

   comparing said speech characteristics with a plurality of voice samples stored for identifying a plurality of callees.

5. The method for identifying a particular callee according to claim 1, further comprising:

   transmitting said voice utterance to a third party device via a network; and

   receiving said callee identity from said third party device.

6. The method for identifying a particular callee according to claim 1, further comprising:

   requesting a voice sample for said particular callee from a third party device accessible via a network; and

   receiving said voice sample for said particular callee for enabling authenticating of said callee identity.

7. The method for identifying a particular callee according to claim 1, further comprising:

transferring said callee identity to an origin device, wherein said origin device is enabled to output said callee identity to a caller, wherein said caller is enabled to select whether to communicate with said callee.

8. The method for identifying a particular callee according to claim 1, further comprising:

receiving a preferred callee selection from a caller at said destination device; and

automatically terminating said call if said callee identity is different than said preferred callee.

9. The method for identifying a particular callee according to claim 1, wherein said destination device is a private exchange network.

10. The method for identifying a particular callee according to claim 1, wherein said destination device is a telephony device.

11. The method for identifying a particular callee according to claim 1, wherein said callee identity comprises at least one from among a callee name, a callee location, a subject of said call, and a device identification.

12. A system for identifying a particular callee, said system comprising:

a destination device enabled to receive a call;

means for detecting a voice utterance of a callee at said destination device; and

means for identifying a callee identity associated with said voice utterance at said destination device.

13. The system for identifying a particular callee according to claim 12, further comprising:

means for prompting said callee, from said destination device, to provide said voice utterance.

14. The system for identifying a particular callee according to claim 12, further comprising:

means for prompting said callee to enter an additional input to verify said callee identity.

15. The system for identifying a particular callee according to claim 12, wherein said means for identifying a callee identity further comprises:

means for extracting speech characteristics from said voice utterance; and

means for comparing said speech characteristics with a plurality of voice samples stored for identifying a plurality of callees.

16. The system for identifying a particular callee according to claim 12, further comprising:

means for transmitting said voice utterance to a third party device via a network; and

means for receiving said callee identity from said third party device.

17. The system for identifying a particular callee according to claim 12, further comprising:

means for requesting a voice sample for said particular callee from a third party device accessible via a network; and

means for receiving said voice sample for said particular callee for enabling authentication of said callee identity.

18. The system for identifying a particular callee according to claim 12, further comprising:

means for transferring said callee identity to an origin device, wherein said origin device is enabled to output said callee identity to a caller, wherein said caller is enabled to select whether to communicate with said callee.

19. The system for identifying a particular callee according to claim 12, further comprising:

means for receiving a preferred callee selection from a caller at said destination device; and

means for automatically terminating said call if said callee identity is different than said preferred callee.

20. The system for identifying a particular callee according to claim 12, wherein said destination device is a private exchange network.

21. The system for identifying a particular callee according to claim 12, wherein said destination device is a telephony device.

22. The system for identifying a particular callee according to claim 12, wherein said callee identity comprises at least one from among a callee name, a callee location, a subject of said call, and a device identification.

23. A computer program product for identifying a particular callee, said computer program product comprising:

a recording medium;

means, recorded on said recording medium, for detecting a voice utterance of a callee at a destination device; and

means, recorded on said recording medium, for identifying a callee identity associated with said voice utterance at said destination device.

24. The computer program product for identifying a particular callee according to claim 23, further comprising:

means, recorded on said recording medium, for prompting said callee to provide said voice utterance from said destination device.

25. The computer program product for identifying a particular callee according to claim 23, further comprising:

means, recorded on said recording medium, for prompting said callee to enter an additional input to verify said callee identity.

26. The computer program product for identifying a particular callee according to claim 23, further comprising:

means, recorded on said recording medium, for extracting speech characteristics from said voice utterance; and

means, recorded on said recording medium, for comparing said speech characteristics with a plurality of voice samples stored for identifying a plurality of callees.

27 The computer program product for identifying a particular callee according to claim 23, further comprising:

means, recorded on said recording medium, for transmitting said voice utterance to a third party device via a network; and

means, recorded on said recording medium, for receiving said callee identity from said third party device.

28. The computer program product for identifying a particular callee according to claim 23, further comprising:

means, recorded on said recording medium, for requesting a voice sample for said particular callee from a third party device accessible via a network; and

means, recorded on said recording medium, for receiving said voice sample for said particular callee for enabling authentication of said callee identity.

29. The computer program product for identifying a particular callee according to claim 23, further comprising:

means, recorded on said recording medium, for transferring said callee identity to an origin device, wherein said origin device is enabled to output said callee

identity to a caller, wherein said caller is enabled to select whether to communicate with said callee.

30. The computer program product for identifying a particular callee according to claim 23, further comprising:

means, recorded on said recording medium, for receiving a preferred callee selection from a caller at said destination device; and

means, recorded on said recording medium, for automatically terminating said call if said callee identity is different than said preferred callee.

35. A method for identifying a callee, said method comprising:

detecting a biometric input at a biometric enabled destination device;

identifying a callee identity associated with said biometric input at said destination device, such that said callee identity is transmittable as an authenticated identity of said callee for a call.

36. The method for identifying a callee according to claim 35, wherein said biometric input comprises at least one from among an eye print, a finger print, a voice input, and a body heat scan.

37. A system for identifying a callee, said system comprising:

a biometric input enabled device;

means for detecting a biometric input at said biometric input enabled destination device;

means for identifying a callee identity associated with said biometric input at said destination device, wherein said callee identity is transmittable as an authenticated identity of said callee for a call.

38. The system for identifying a callee according to claim 37, wherein said biometric input comprises at least one from among an eye print, a finger print, a voice input, and a body heat scan.

39. A computer program product for identifying a callee, said computer program product comprising:

a recording medium;

means, recorded on said recording medium, for detecting a biometric input at said biometric input enabled destination device;

means, recorded on said recording medium, for identifying a callee identity associated with said biometric input at said destination device, wherein said callee identity is transmittable as an authenticated identity of said callee for a call.

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Gallick

(10) Patent No.: **US 6,678,359 B1**
(45) Date of Patent: **Jan. 13, 2004**

(54) **CALLED PARTY IDENTIFICATION IN PACKET SWITCHED NETWORKS**

(75) Inventor: Robert Lawrence Gallick, Phoenix, AZ (US)

(73) Assignee: AG Communication Systems Corporation, Phoenix, AZ (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: 09/544,181

(22) Filed: Apr. 6, 2000

(51) Int. Cl.^{7} H04M 1/64

(52) U.S. Cl. 379/88.17; 379/88.2

(58) Field of Search 379/210.02, 88.11, 379/88.17, 88.2, 93.01, 201.01, 207.15; 370/352–356

(56) **References Cited**

U.S. PATENT DOCUMENTS

6,477,246 B1 * 11/2002 Dolan et al. 379/211.02

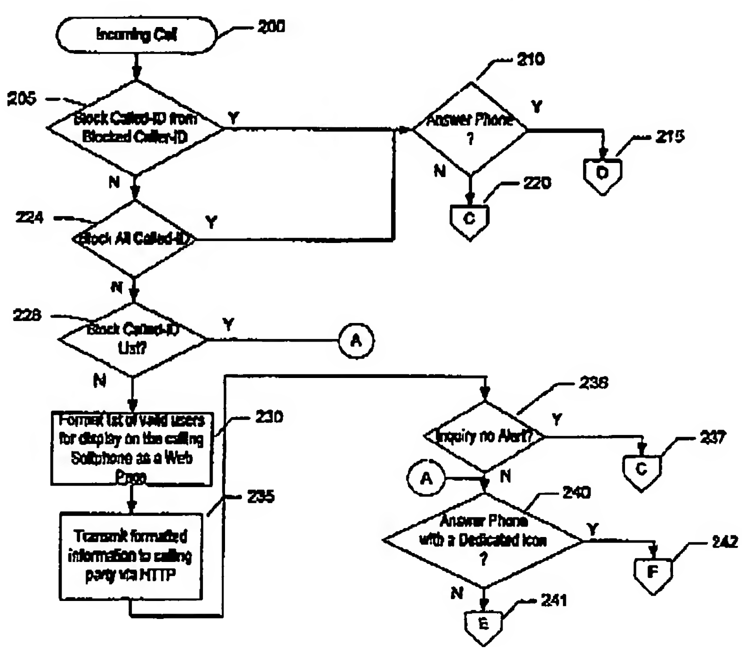
* cited by examiner

Primary Examiner—Creighton Smith

(57) **ABSTRACT**

In processing of VoIP calls, a called facility, prior to the cut through of ringing that would alert personnel to the presence of an incoming call, transmits previously stored information that identifies the persons at the called facility that are authorized to handle particular types of calls. In response to the receipt of caller-ID information, the called facility queries a database to determine whether (a) the caller is known and, if so, whether there are particular call handling instructions stored for this caller, and (b) whether the called facility is classified as blocked from participating in called party identification service. If the database response indicates that the called facility participates in called party identification service, the data base sends to the caller, before the call is answered, a list of persons at the called facility who are authorized to answer the call and, advantageously, some indication of what that persons area of responsibility may be. When one of the authorized persons answers the call a brief selection message is transmitted to the caller identifying which of the authorized persons is answering the call. For example, the person answering the call may simply press a button at his telephone corresponding to his name or, in an alternative embodiment, a speech sample from the answering person is analyzed and the selection message is sent based on the speech analysis. In the event that the call is not answered, the caller may, after reviewing the area of responsibility information on the list, select the most appropriate persons on the list to receive a detailed message.

9 Claims, 5 Drawing Sheets



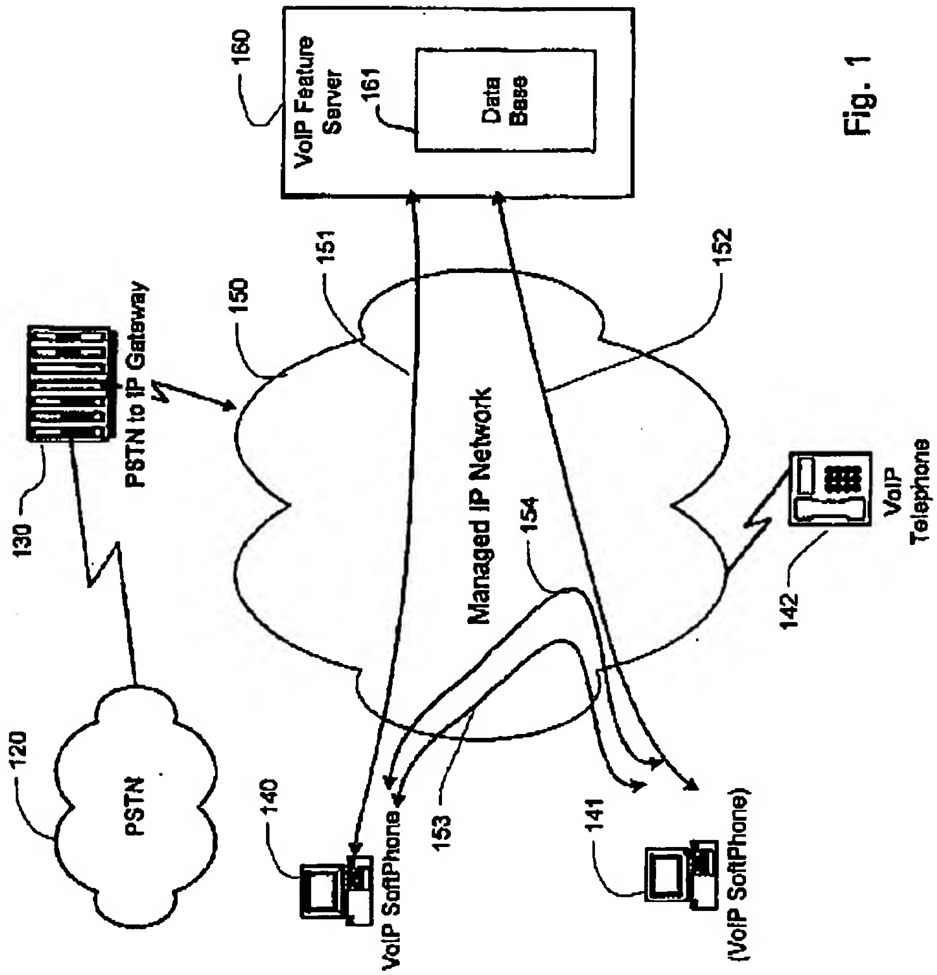

VoIP Feature Server
160
161
Data Base
151
152
150
PSTN to IP Gateway
130
Managed IP Network
154
VoIP Telephone
142
120
PSTN
140
VoIP SoftPhone
153
141
(VoIP SoftPhone)

Fig. 1

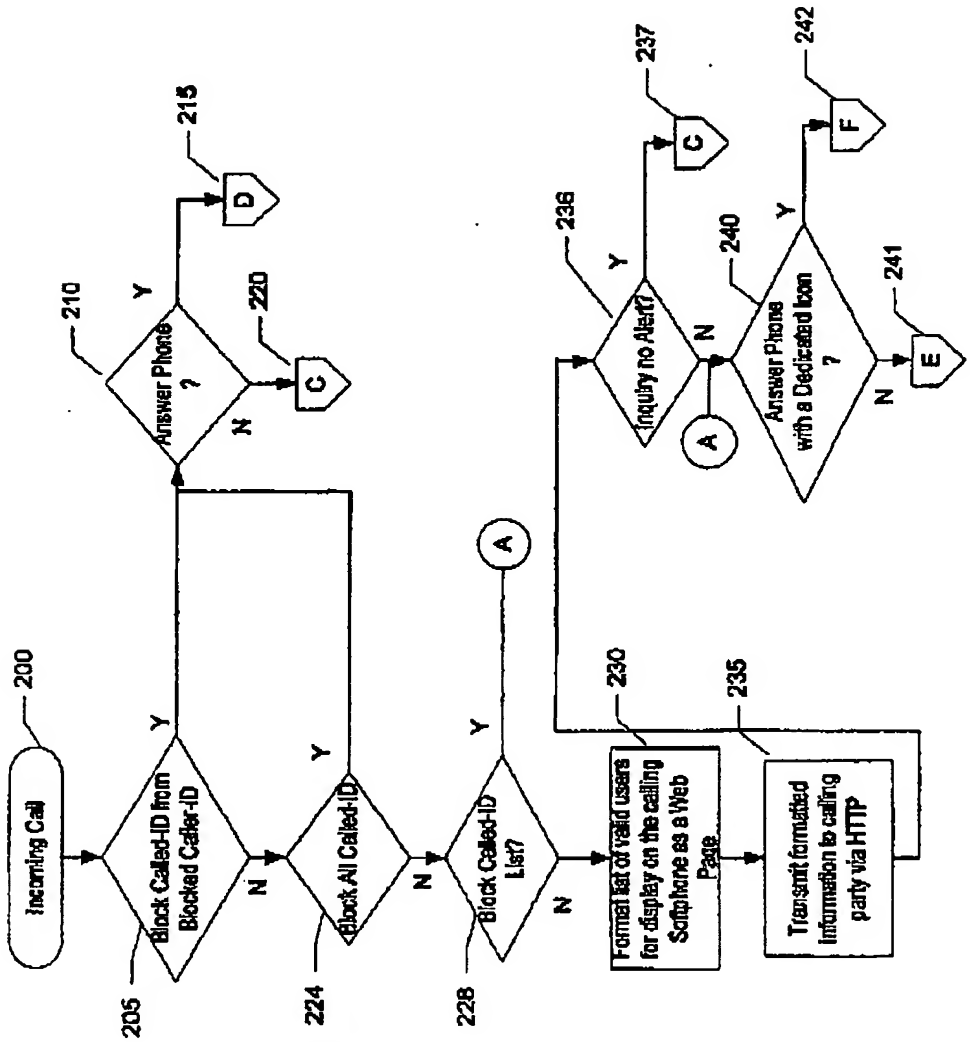
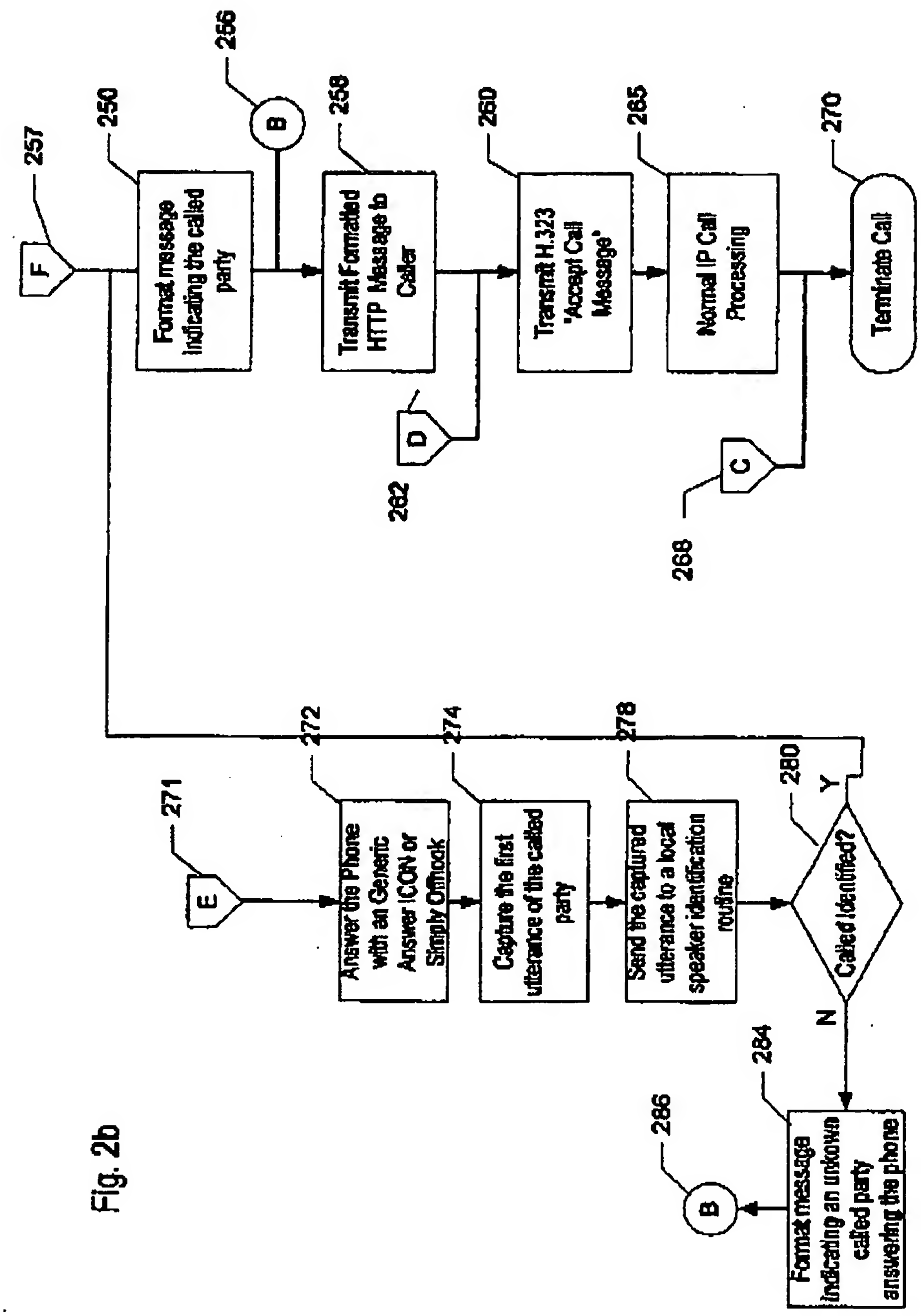

Incoming Call
200
Block Called-ID from Blocked Caller-ID
205
Block All Called-ID
224
Block Called-ID List?
228
Format list of valid users for display on the calling Softphone as a Web Page
230
Transmit formatted information to calling party via HTTP
235
Answer Phone ?
210
D
215
C
220
A
Inquiry no Alert?
236
C
237
Answer Phone with a Dedicated Icon ?
240
F
242
E
241
Y
N

Fig. 2a



F
257
Format message indicating the called party
250
B
256
Transmit Formatted HTTP Message to Caller
258
Transmit H.323 "Accept Call Message"
260
Normal IP Call Processing
265
Terminate Call
270
D
282
C
268
E
271
Answer the Phone with an Generic Answer ICON or Simply Offhook
272
Capture the first utterance of the called party
274
Send the captured utterance to a local speaker identification routine
278
Called Identified?
280
Y
N
Format message indicating an unkown called party answering the phone
284
B
286

Fig. 2b

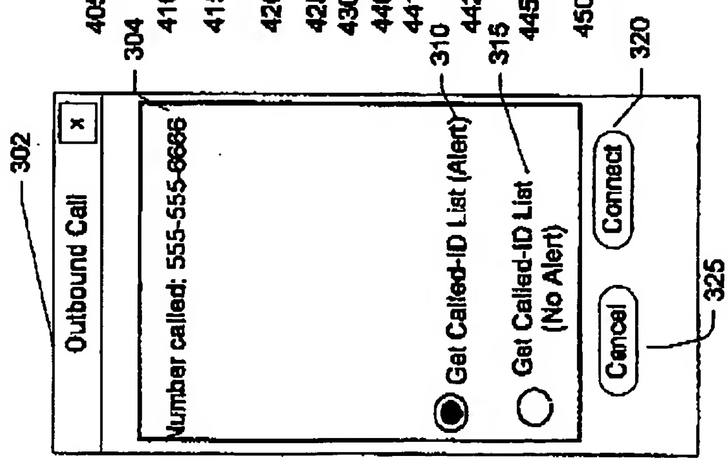


Fig. 3
Calling Party Screen During Out-bound Call

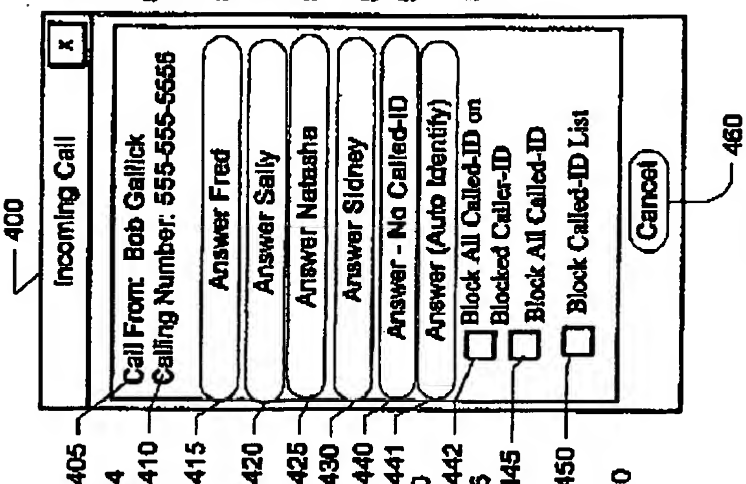


Fig. 4
Called Party Incoming Call Screen

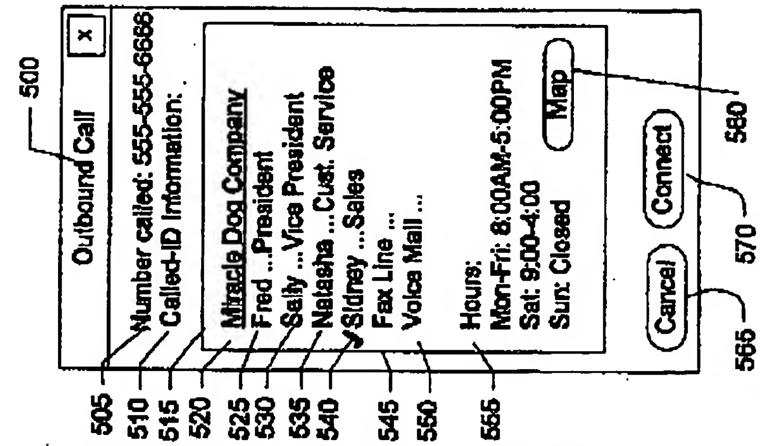


Fig. 5
Calling Party Screen After Receipt of Called-ID Information

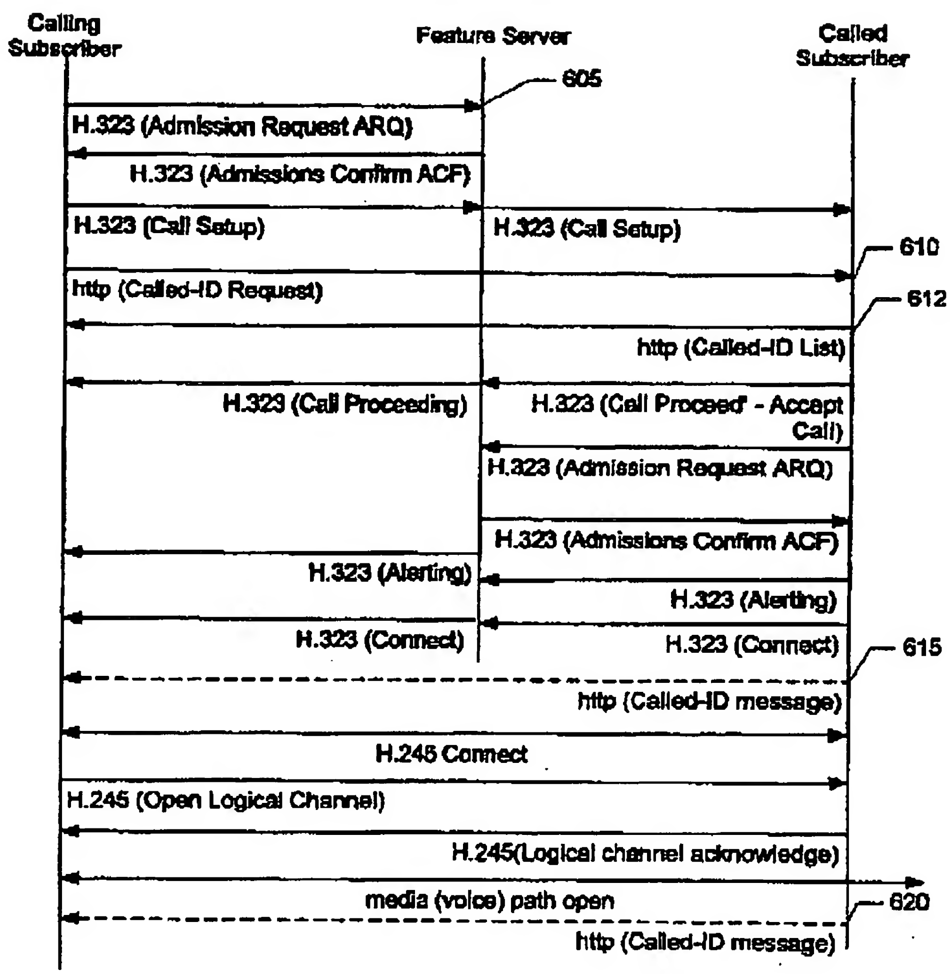

Calling Subscriber
Feature Server
Called Subscriber
605
H.323 (Admission Request ARQ)
H.323 (Admissions Confirm ACF)
H.323 (Call Setup)
H.323 (Call Setup)
610
http (Called-ID Request)
612
http (Called-ID List)
H.323 (Call Proceeding)
H.323 (Call Proceed - Accept Call)
H.323 (Admission Request ARQ)
H.323 (Admissions Confirm ACF)
H.323 (Alerting)
H.323 (Alerting)
H.323 (Connect)
H.323 (Connect)
615
http (Called-ID message)
H.245 Connect
H.245 (Open Logical Channel)
H.245(Logical channel acknowledge)
media (voice) path open
620
http (Called-ID message)

Fig. 6

CALLED PARTY IDENTIFICATION IN PACKET SWITCHED NETWORKS

FIELD OF THE INVENTION

This invention relates to the identification of users of communications systems and, more particularly, to identification of the party answering a voice over internet telephone (VoIP) telephone call.

BACKGROUND OF THE INVENTION

In circuit switched telephony it is comparatively easy to identify a calling party's terminal by mapping the physical, central office location of the telephone line serving the terminal to a directory number. Once the calling terminal's directory number has been obtained, it may be forwarded to the called party, advantageously prior to answer or during a silent interval of ringing. In the prior art, voice over internet (VoIP) telephony which is transmitted over a packet switched network such as the Internet, software called "softphone" software is typically employed that conforms to the protocols defined in Recommendation H.323 of the International Telecommunications Union (ITU) entitled "Visual Telephone Systems and Equipment for Local Area Networks which Provide a Non-guaranteed Quality of Service". Neither circuit switched nor packet switched telephony, however, provides any means for identifying the party who answers the call. When the called telephone number is directed to a facility at which any of several individuals (e.g., family members or the employees of a small business) may answer a call it would be desirable to provide the caller with some indication of the identity of the person who actually answers the call. This would have several advantages. From the caller's perspective, it is desirable to know, without asking, who the person is who answers the call. From the standpoint of a business, providing the caller the name of the person who answers the call may dispel some of the aura of "faceless" communication that calls to some businesses seem to acquire and instead help the business to project an image of "personal service". Thus, it would be advantageous to provide the caller with the name of the person who actually answers the call. It would also be advantageous to allow the caller to leave a message for a particular person at a called facility, even when the caller has no knowledge before making the call who that person might be.

SUMMARY OF THE INVENTION

In accordance with the principles of the invention, the above problems are solved in an arrangement particularly adapted to VoIP calls, that provides Called Party Identification, i.e., which allows a called facility to provide the caller with information that identifies the person who answers the call without the need of the caller to ask and so that the answering individual may be distinguished from any other individual who may answer the call at the facility. According to one aspect of the invention, before an incoming call is answered, the called facility, if it receives caller-ID information, queries a database to determine whether (a) the caller is known and, if so, whether there are particular call handling instructions stored for this caller; and (b) whether the called facility is classified as blocked from participating in called party identification service. If the database response indicates that the called facility participates in called party identification service, the data base sends to the caller, before the call is answered, a list of persons at the called facility who are authorized to answer the call and, advantageously, some indication of what that persons area of responsibility may be. When one of the authorized persons answers the call a brief selection message is transmitted to the caller identifying which of the authorized persons is answering the call. For example, the person answering the call may simply press a button at his telephone corresponding to his name or, in an alternative embodiment, ~~speech sample from the answering persons is analyzed and the selection message content based on the speech analysis~~. In the event that the call is not answered, the caller may, after reviewing the area of responsibility information on the list, select the most appropriate persons on the list to receive a detailed message.

BRIEF DESCRIPTION OF THE DRAWINGS

The foregoing and other features of the invention may become more apparent from a reading of the ensuing description together with the drawing, in which:

FIG. 1 is a functional block diagram illustrating a communications system which is arranged to provide called party identification service in accordance with the present invention;

FIGS. 2a and 2b taken together are a flow chart showing the processing of a call incoming to the called facility;

FIG. 3 depicts the calling party screen during an outgoing call;

FIG. 4 depicts the called party screen prior to receipt of an incoming call;

FIG. 5 depicts the calling party screen after the receipt of a called-ID message; and

FIG. 6 is a chart of messages passed in performing called-ID.

DESCRIPTION

FIG. 1 shows a packet switched IP network 150 to which a plurality of computer terminals 140, 141 running voice over IP (VoIP) "softphone" software and/or dedicated VoIP telephones 142 have access. Packet network 150 may include the functionality of the well-known Internet protocol and may be public or be a private, i.e., managed access, packet network. Access between IP network 150 and the public switched telephone network (PSTN) 120 may be effected via a conventional PSTN to IP gateway 130. Communications connections among computer terminals 140, 141 and VoIP telephones 170, 171 and connections between them and gateway 130 is regulated by feature server 160 which performs gatekeeper functions. Feature server 160 may advantageously employ the functionality of the feature server disclosed in co-pending application Ser. No. 09/511257, filed Feb. 23, 2000 entitled "Questionnaire Based Call Routing".

A conventional VoIP call from one terminal such as 140 to another terminal such as 141 would begin (assuming the aforementioned H.323 protocol is employed), with the calling terminal sending a limited bandwidth call control message 605 (see FIG. 6) called an admission request signal (ARQ) to the system gatekeeper (or, in the present application, a feature server 160 which acts as a VoIP gatekeeper). It should be apparent, however, that while the H.323 protocol is referred to in describing the illustrative embodiment, any similar VoIP protocol could be used. If the system gate keeper accepts the ARQ message, the gatekeeper confirms its acceptance by returning to the calling terminal an Admissions Confirm (ACF) call control

message, followed by the gatekeeper providing the calling terminal with the called terminal's IP address. This is sufficient for the network to establish an end-to-end connection 153 between the calling and called terminals over which the bandwidth is limited only by the capacity of the network.

In accordance with the invention, the caller and called terminals are provided with graphical user interface (GUI) displays that allow the called facility to provide the caller with information and options useful in allowing the caller to designate how the call should be handled at the called facility. These GUIs are shown in FIGS. 3 through 5. In FIG. 3 the calling subscriber's softphone screen display is making an outbound call 302. The called PSTN number or IP address is indicated at 304. The display is provided with two option buttons, 310 and 315. If "Get called-ID List (No Alert)" is selected in 315, then a request will be made to the called facility to retrieve a list of personnel authorized to handle the various types of inquiries callers may make and to provide this list, and any other useful information, such as hours open for business, to the caller without alerting personnel at the called facility to the presence of the incoming call by an audible tone or an incoming message display at the called terminal. Alternatively, conventional alerting of the called party can be specified by the caller selecting button 310.

FIG. 4 illustrates the GUI screen display presented to the called facility. At 400 the presence of an incoming call is indicated. Caller-ID information, obtained from the H.323 setup message, is displayed at 405 and 410. A number of selectable icons are displayed representing any of the individuals at the called facility that are authorized to answer particular incoming call inquiries. These icons would be pre-programmed by the called subscriber prior to placing the system in operation to handle incoming calls. A number of icons illustrative of the type of information that would be useful to the caller are shown, including: "Answer Fred" 415, "Answer Sally" 420, "Answer Natasha" 425, and "Answer Sidney" 430. In addition, icons showing "Answer—No called-ID" 440 and "Answer (Auto Identify)" 441 are shown. When one of these icons is selected by the individual answering the call at the called facility, a message is sent to the caller identifying the answering individual.

If the call happens to be answered by an individual at the called facility whose name does not correspond to one of the available icons, button 441 can be selected in which event the call will be answered and the first utterances of the called party will be captured and analyzed to determine of the voice sample corresponds to one on file. If the called individual is identified, an appropriately formatted message will be transmitted to the calling subscriber. If the individual cannot be identified, an appropriate message indicating that the called party could not be identified is sent to the calling party.

The next option on the called facility GUI display is box 442, "Block All called-ID on Blocked Caller-ID". When this check box is checked, if the incoming call has not transmitted caller-id information in the setup message, no called-ID is transmitted back to the calling party whether or not caller-ID information has been transmitted. The next option shown is "Block All called-ID", box 445. If this option is selected, no called-Id information will be transmitted back to the calling party. If box 450, "Block called-ID List" is checked, the list of authorized individuals at the called facility will not be transmitted. If, however, the individual answering the call at the called facility has been identified that identity will be transmitted back to the calling subscriber. It should be obvious, that other derivative options could be created such as Block called-ID list to a particular list of callers to provide selective blocking of called-id to the calling party.

FIG. 5 shows the GUI screen displayed to the calling subscriber after the called-ID information has been transmitted to the caller containing additional useful information. 500 shows "Outbound Call" and the called number is shown at 505. A new label 510 appears indicating that the rest of the displayed information pertains to the called facility. 515 shows an instance of a "web" page window transmitted by the called facility to the caller. For example, the business name of the called facility, in this case, "Miracle Dog Company" appears at 520. The names and responsibilities of personal at the called facility authorized to handle particular types of inquiries are displayed at 525, 530, 535, and 540. The check mark 540 next to "Sidney . . . Sales" indicates that the call has been answered by an individual named "Sidney". In addition, the FAX line number of the called facility is displayed at 545 and a voice mail number is displayed at 550. The hours of operation of the called facility's business is displayed at 555 and an additional hyperlink 560 is displayed to provide the map location of the business. Since the display has been formatted as a web page by the called subscriber any additional information desired may be transmitted via hyperlinks.

It should be noted that all of the information indicated (except the identity of the individual who actually answers the call) is advantageously transmitted to the caller before personnel at the called facility need be notified of the presence of the incoming call and without the need of operator or telephone attendant intervention.

For completeness, icons depicting "Cancel" the call or "Connect" the call to the selected individual in the called-ID list is shown in 565 and 570, respectively. With the GUI displays of FIGS. 3–5 the caller may select the individual at the called facility with whom they which to speak and then, by selecting the "connect" button 570 cause the called facility to alert the indicated individual to the presence of the incoming call.

The sequence of operations initiated by the use of the option buttons in the GUI displays of FIGS. 3–5 are depicted in the flow diagram of FIGS. 2a and 2b. Processing of an incoming call begins at 200. At 205 a database lookup occurs based on incoming caller-ID information transmitted from the calling party. Such a database may be maintained locally or advantageously may be included in the functionality of feature server 160. For purposes of simplified illustration, the database will be deemed to be included in feature server 160 as database 161. However, it should be apparent that such a data base could be maintained on a terminal running a softphone software application.

It will be assumed that the customer at the called facility desires to have information pertinent to the called facility (e.g., the list of personnel authorized to handle specific types of calls, the hours that the facility is open for business, etc.) that has been loaded in the data base provided to the caller before the call is answered. The caller may then select a person from the list to direct the call to that person. When the call is answered, the person answering the call at the called facility presses a button at his terminal to identify himself to the caller. However, the customer may know in advance that certain callers do not need this information and, accordingly, database 161 stores a table listing calling party dialed numbers (DNs) or IP addresses for which called party identification information (called party-ID) is blocked.

From the incoming caller-ID information, table lookup determines whether the caller is one to whom called_ID

information will not be transmitted. If table lookup indicates that called party-ID is blocked for this incoming call, control advances to step 210 where the called terminal will be alerted. If the call is not answered, control passes to step 220 where the call is terminated. If the called party answers the call, control proceeds to step D 262 (FIG. 2b) where an H.323 "Accept Call" message 260 is transmitted to the calling party and conventional IP call processing steps 265 take place pertaining to VoIP call setup and connection which need not be detailed here. At the conclusion of the telephone call, the call is terminated at 270.

If table lookup at step 205 indicates that called-ID is not blocked for this caller, processing continues to step 224 to determine if the customer at the called facility has marked box 445 on the terminal display, FIG. 4, to block all called-ID. If "Block All called-ID" has been selected by the called subscriber, then the logic follows the same path as was previously described for Block called-ID on Blocked Caller-ID.

Processing continues to step 228 in FIG. 2a which checks whether the party at the called facility had checked screen display box 450, FIG. 4 "Block called-ID List". When box 450 has been checked, no information pertaining to the called subscriber is transmitted to the calling subscriber prior to the alerting of the called subscriber. When the called subscriber answers after being alerted, step 240, and is successfully identified, the identification is forwarded to the caller.

If box 450 FIG. 4 had not been checked, processing in FIG. 2a continues to step 230 which formats the information that the customer at the called facility desires to be presented to the caller. This information may advantageously be formatted using http protocol in the same way that a "Web" page display is formatted for transmission over an IP network and is transmitted to the caller's terminal at step 235. The information, see FIG. 5, may advantageously include the business name of the called facility, i.e., "Miracle Dog Company" 520; a list of personnel authorized to answer the call and their position 525–540; hours of business 555; hypertext links to additional information, such as "map" 560; as well as any text, graphic, or multimedia information, which may be commonly associated with a "web" page. In addition to this information, the terminal display effected at the caller's terminal is presented with some "radio" button options to allow the caller to control the call, as will hereinafter be explained.

Processing in FIG. 2a then proceeds to step 236. At this point, the program determines if the calling subscriber had selected one of the buttons (e.g., 315, FIG. 3) indicating a desire not to alert the called subscriber until the information that had been transmitted in step 235 has been studied. If this option has been selected, the program jumps to off-page connector C to indicate that processing continues in FIG. 2b where the call would simply terminate using conventional H.323 call termination. If the caller has selected this option, the caller can later cause the call to be completed by pressing the "connect" button 320, FIG. 3.

If the answer to step 236 is "no", the logic proceeds to 240 in FIG. 2a and the called subscriber is alerted to the incoming call. If the called subscriber answers the phone by selecting one of the dedicated icons, such as those shown in FIG. 4 at 415, 420, 425, or 430, the logic continues to off-page connector F and to FIG. 2b where a message is formatted at step 250 with the name associated with the selected icon, such as "Sidney". The message, which is actually part of a Web page, is transmitted via http at step 258. The call would proceed as normal, and would terminate on disconnect at 270 FIG. 2b.

The user at the called facility has the option to answer the call without selecting one of the dedicated icons by selecting option button 440 "Answer—Auto Identify" in which case processing continues at E, FIG. 2b. At this point the first utterance of the answering called party is captured and voice identification software analyzes the speech to identify the person answering the call at the called facility. For example the called subscriber may answer "Miracle Dog Company" or simply "Hello". This phrase would be captured and in 27[illegible] and sent to a local voice identification recognizer located on the local personal computer where the softphone resides o[illegible] ~~[illegible]~~ If the voice of the party answering is identified at step 280, a message is formatted at step 250 which identifies the answering party ~~and the formatted message [illegible] the caller via HTTP in step 258~~. Thereafter, the call proceeds as normal and, on disconnect, the call would terminate at 270.

If the answering party was not identified at step 280, a message indicating an "Answering Party Could not be identified" would be formatted at step 284. The logic continues to on-page connector B where the message would be transmitted via HTTP in 258. Call processing would continue as has been previously described.

FIG. 6 shows the message flow for the called-ID implementation. After the H.323 admissions confirmation in 605, feature server 160 sends the IP address to the calling party. At that point, a normal H.323 call setup would continue with messages being sent from the calling party to the called subscriber. The setup up will contain the caller-id (i.e. name) of the calling party as well as the calling subscribers' IP address. Concurrently, in 610 a request for called-ID information is sent from the calling subscriber to the called subscriber. This message will take on the form: http://called_party_IP_address/feature.asp?Command=called_ID. In response, the called subscribers' softphone transmits a web page to the calling subscriber's softphone containing the called-ID information. The list is not transmitted if called-ID is blocked by the called party. If the called-ID subscriber answers the phone by pressing a unique icon identifying themselves (such as Sidney), then another out of band message can be sent to the calling party before the audio path is setup as is shown in 615. This message would be an HTTP message with the data "Sidney" encapsulated in it. ~~If the subscriber chooses to answer the phone and speaking—allowing the speaker identification software to identify them, then the audio path would proceed to be connected and the message shown in 615 would not be sent.~~ Instead, the first utterance of the called party would be captured and sent to caller verification routines for an attempted identification of the called party. If successful, an out of band message would be sent at 620 back to the calling party indicating the party speaking at the called telephone instrument. In this case, "Sidney". If the called party could not be identified by the speaker identification software, a message formatted with "Unidentified Caller" or similar would be transmitted out of band to the calling party.

If the calling party was determined to be calling from a handheld device such as a cellular phone with a display, the program at the called facility or feature server will transmit the information to the calling instrument in a format capable of being displayed on such a device such as Handheld Device Markup Language (HDML). (The type of device being used by the caller is one of the items of information furnished in accordance with H.323 protocol.) It should also

be evident that this called-ID information may be transmitted to calling telephones served by the PSTN where the calling party has an Analog Display Services Interface (ADSI) or other display device with similar capabilities, in which case the information would be transmitted as an FSK message. It would also have to filter to the content from that of a graphical nature to that of a text display. Further and other modifications will be apparent to those skilled in the art and may be made without, however, departing from the spirit and scope of the invention.

What is claimed is:

1. A method of processing voice over internet protocol calls, comprising the steps of:

storing in a database graphically encoded information pertaining to individuals at a called facility who are authorized to handle different types of calls directed to said facility;

querying said database in response to a call incoming to said facility;

transmitting said graphically encoded information to the caller's terminal associated with said call in order to enable said caller to make a selection among said individuals; and

further processing said call in accordance with said selection thereafter made by said caller.

2. A method of processing calls according to claim 1 wherein said graphically encoded information is formatted for transmission to said caller's terminal as a web page.

3. A method of processing calls according to claim 1 wherein said graphically encoded information is transmitted prior to alerting personnel at said called facility to the presence of said incoming call.

4. A method of processing calls according to claim 3 wherein said information includes a plurality of icons identifying said authorized individuals.

5. A method of processing calls according to claim 4 wherein a person answering a call at said facility is identified and wherein one of said plurality of icons transmitted to said caller's terminal is illuminated corresponding to said identified person.

6. A method of processing calls according to claim 5 wherein said person answering a call at said facility is identified by voice analysis.

7. A method of processing calls according to claim 6 wherein said plurality of icons is displayed to a person answering a call at said facility for selection of an icon to be illuminated at said caller's terminal.

8. A method of processing calls according to claim 3 wherein said further processing includes receiving from a caller a voice message to be stored for individual at said facility designated by said selection.

9. A method of processing calls according to claim 3 further including means for storing an indication to block transmission of said stored graphical information to particular callers.

* * * * *

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(54) METHOD AND SYSTEM OF OFFERING WIRELESS TELECOMMUNICATION SERVICES IN A VISITED TELECOMMUNICATION NETWORK

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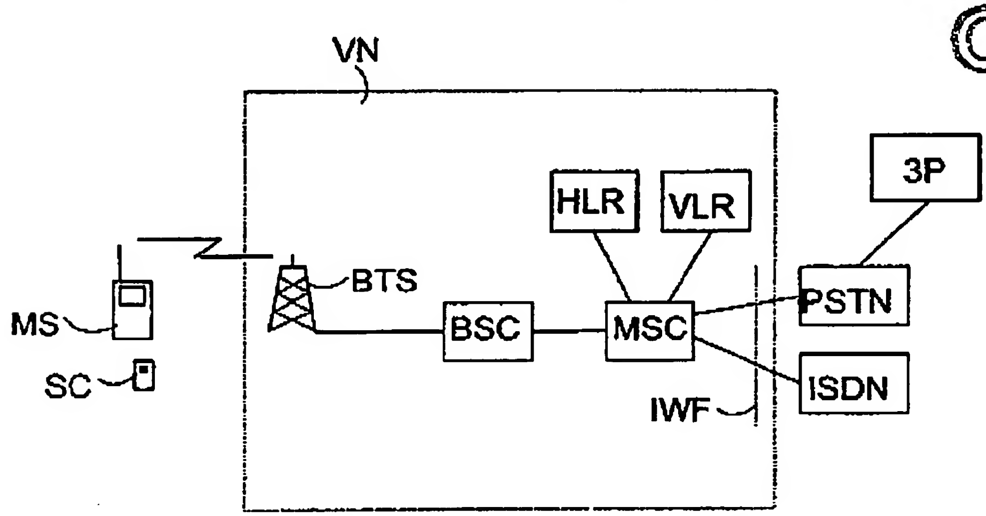
(57) ABSTRACT

A method and system of offering wireless telecommunication services in a visited telecommunication network for a terminal device which does not have a first identifier accepted by the network. In the method, the visited network gives the terminal device a right to a temporary use of a telecommunication connection and establishes a connection with a third party in order to obtain a confirmation of paid services, for example. The third party checks a second identifier transmitted by the terminal device and transmits the confirmation to the visited network, which offers telecommunication services for the terminal device.



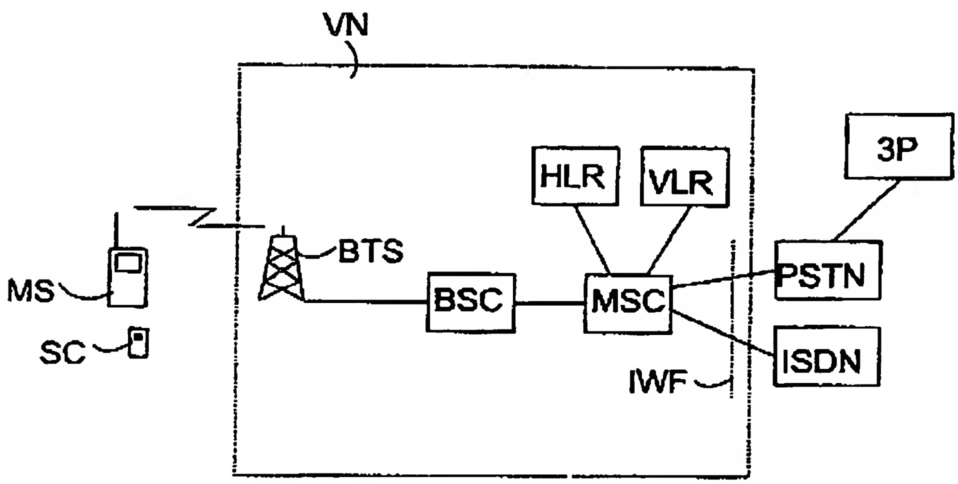

VN
3P
HLR
VLR
BTS
MS
BSC
MSC
PSTN
SC
ISDN
IWF

Fig. 1

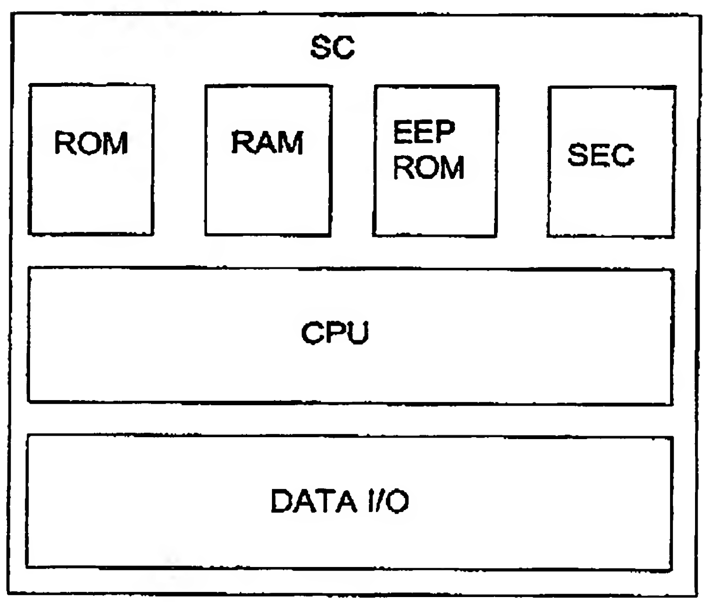
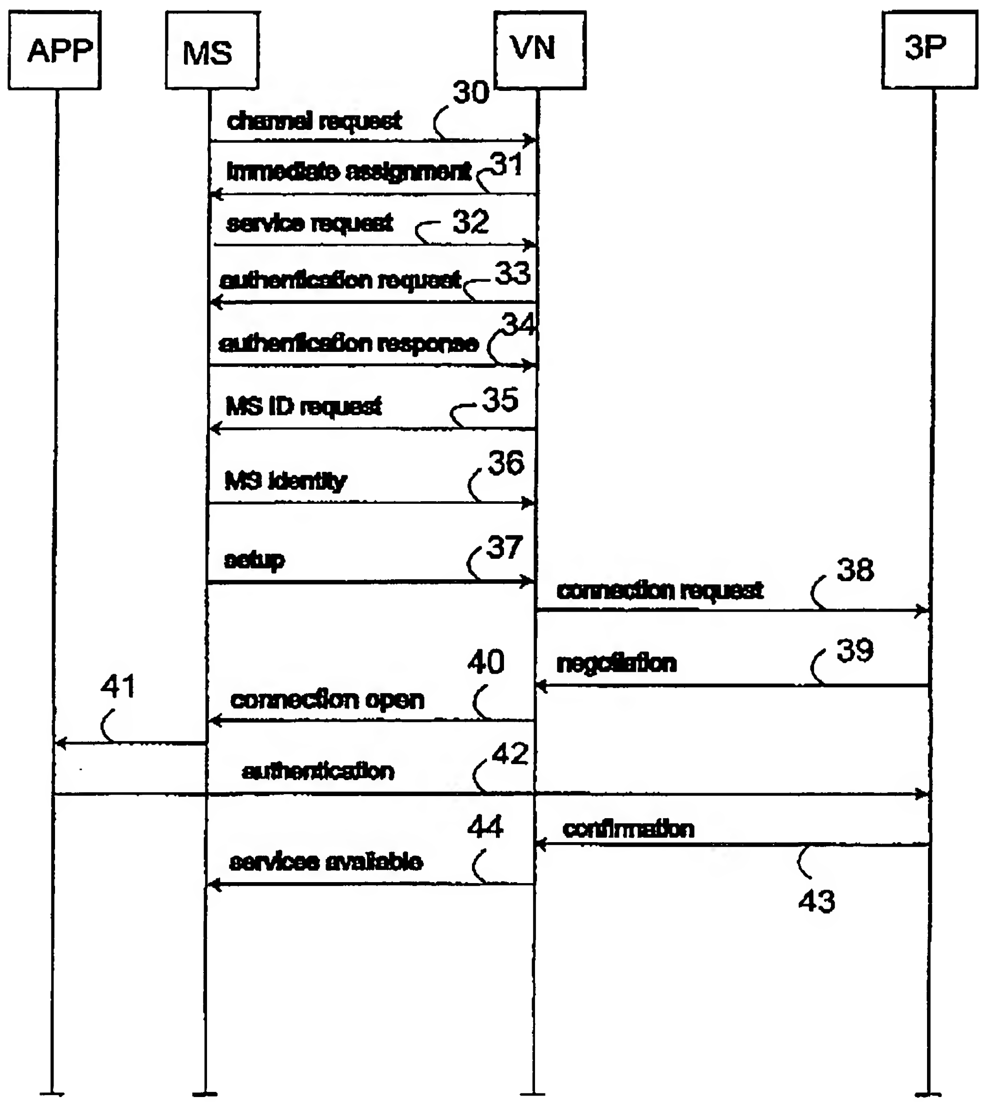

SC
ROM
RAM
EEP ROM
SEC
CPU
DATA I/O

Fig. 2



APP
MS
VN
3P
30
channel request
immediate assignment
31
service request
32
authentication request
33
authentication response
34
MS ID request
35
MS identity
36
setup
37
connection request
38
negotiation
39
40
connection open
41
authentication
42
confirmation
44
services available
43

Fig. 3

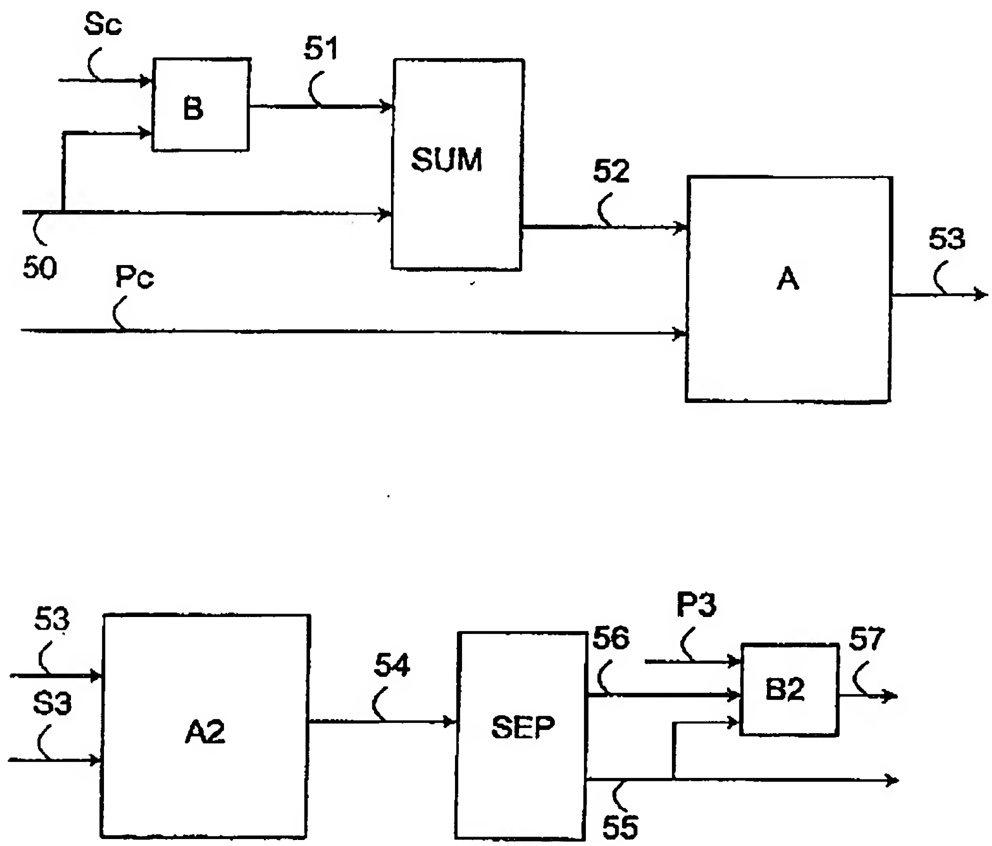


Fig. 4

METHOD AND SYSTEM OF OFFERING WIRELESS TELECOMMUNICATION SERVICES IN A VISITED TELECOMMUNICATION NETWORK

[0001] This application is a Continuation of International Application PCT/FI00/00429 filed May 12, 2000 which designated the U.S. and was published under PCT Article 21(2) in English.

[0002] The invention relates to telecommunication service provision in a visited telecommunication system.

[0003] In mobile communication networks, a transmission path consists at least partly of a wireless section, whereby data is transmitted via the radio path. The radio path is a resource, which is physically open and which involves security risks. In digital mobile communication networks, various solutions for improving the security of data transmission, e.g. methods of encryption and user identification, have been developed. As an example, this application uses the second generation mobile communication system, the GSM system, in which a data transmission encryption can be used that is difficult to decrypt, whereby speech converted into a digital form and a data signal are encrypted, i.e. coded in a mobile station to be transferred over the radio path. Correspondingly, the encrypted transmission received in the GSM network is decoded into unencrypted speech and data. The encryption and user identification utilize encryption keys and algorithms, which are preferably only available for appropriate transmission and reception means.

[0004] To prevent misuses and secure charging in particular, a user identification is arranged in the GSM system. To profit from the services provided by the GSM system, a subscriber, which may be a different person than the real user of the mobile station, has to make an agreement with the operator possessing the network. To identify the user, for instance, the operator gives the mobile station user a so-called smart card comprising a SIM application (Subscriber Identity Module). In this application, a SIM card is regarded as a user-specific identifier, whereby the user is also a subscriber. The SIM card can naturally also be considered as a subscriber-specific identifier. Smart cards, such as a SIM card, comprise at least a microprocessor and memory. The user identification is typically arranged in smart cards by means of a PIN identifier (Personal Identity Number), and so the card can only be used by the user who knows the PIN.

[0005] SIM cards, too, use a PIN code to check the right user. Using information of the SIM card in a GSM connection set-up, a mobile station transmits the GSM network identification data, on the basis of which the SIM card and the user are identified. The SIM card particularly comprises information concerning the mobile operator, e.g. SIM card-specific user identifier IMSI (International Mobile Subscriber Identity) of the mobile communication services user and the temporary identifier of the location area TMSI (Temporary Mobile Subscriber Identity). The more detailed specifications of the SIM card are described in the GSM standard 11.11.

[0006] As a mobile station sets up a connection with the GSM network, the mobile communication network checks the TMSI, which the mobile communication network has allowed to be used instead of the IMSI, if the mobile station has been in the location area of the network in question last time when the connection was set up. If no TMSI is available, the mobile communication network requests the mobile station for the IMSI, which the mobile communication network checks from the home location register HLR. There-after, an authentication is typically performed, whereby it is checked that the SIM card is right and that it relates to the IMSI. The mobile station further comprises an identifier for the mobile station IMEI (International Mobile Equipment Identity), which can be used for checking, which mobile station is using the mobile communication network. A more detailed description of the GSM system can be found for example in the work "The GSM System for Mobile Communications",M. Mouly and M. Pautet, Palaiseau, France, 1992, ISBN:2-9507190-0-.

[0007] In this application, a home network refers to a mobile communication network, with which a mobile station user has an agreement, to which he has a right to access and from which he has received means for user identification in the GSM networks (i.e. a SIM card in the GSM system). A user has always a direct right to access to the home network. A visited network refers to a mobile communication network, to which a mobile station user has no direct right to access beforehand. When a mobile station is roaming, it is in some other area than that of the home network, i.e. in the area of a visited network. However, if a SIM card has been inserted into the mobile station, the visited network can, on the basis of the IMSI, contact the home network, which identifies the user, and the mobile station is offered services in the visited network, which means the user has a so-called indirect right to access to the network. This requires, however, that an agreement is made beforehand between the visited network and home network and that the telecommunication connections between them are working. The user has no right to access for example, when there is no functional SIM card in the mobile station or no roaming agreement exists between the visited network and home network. A mobile station without right to access refers in this application to a mobile station, whose user does not have a right to use telecommunication services of a visited network.

[0008] The use of GSM telecommunication services according to the GSM standards requires principally that a SIM card is inserted into the mobile station. As an exception to this, emergency calls can be made from various GSM mobile communication networks without a SIM card. In an emergency call, it is possible to contact the emergency number without that the user's SIM card would be identified. The emergency call solution can only be utilized when the call is made to specific emergency numbers.

[0009] However, all mobile station users do not want to commit themselves to services provided by a specific mobile communication operator, but they possibly want to select the operator they use at each time. People who use a mobile station seldom do not always want to make a special agreement, and many of them do not want to give their personalia to the mobile communication network. Because of charging problems, customers' creditworthiness is checked in many countries, before making mobile communication service agreements. Different GSM operators have offered various prepaid SIM cards, which can be bought like conventional telephone cards. Prepaid SIM cards can be used in the same way as conventional SIM cards; the

difference is that telecommunication services have been paid for in advance. These prepaid SIM cards have proved to be very popular.

[0010] Smart cards have recently become more common as means of payment in particular. Instead of magnetic tape cards, for example, smart cards have been introduced in bank cards. Smart cards are planned to be used for securing electronic commerce via the Internet, for example. A smart card reader is connected to a computer, and using the information of the smart card, a user identification and encryption of a credit card number, for example, are performed. As the use of general-purpose payment applications based on a smart card become more popular, many mobile station users may want to use them for paying for mobile communication services as well.

[0011] The patent application WO 9834430 describes a method of allocating a temporary username from a wireless telecommunication network. In said publication, a mobile communication service is described, which is used without a previously made agreement and without a SIM card. According to the central idea of the publication, a mobile communication network can thus be contacted anonymously, and the network gives a temporary username for the duration of the call. By means of this identifier, the service offered for a specific mobile station is distinguished from the services of other mobile stations. This makes the use of mobile communication services more flexible and provides the user with more alternatives. The method described in the publication may well be used in free mobile communication services and also in the method of payment on a smart card, on which the mobile communication operator can rely.

[0012] A problem in the above arrangement is that it is not possible for the network operator to identify a user without identification means, such as a SIM card, and the payer of the bill, for example, cannot thus be guaranteed. The network operator has no guarantee of obtaining a compensation for the use of the telecommunication connection, unless some other reliable credit card is simultaneously charged. If the user remains unidentified, there is a growing risk of potential criminal actions, since the only identifier that is required is a mobile station identifier. According to the prior art, it is not possible to contact a visited mobile communication network, with which the home network does not have a roaming agreement.

[0013] The object of the invention is thus to provide a method and an apparatus for implementing the method such that the above problems can be avoided. The object of the invention are achieved with a method, a telecommunication system, and a network element which are characterized by what is disclosed in the independent claims. The preferred embodiments of the invention are disclosed in the dependent claims.

[0014] The method comprises the following steps: A connection is established from a terminal device to a fixed network providing network access for the terminal device. A first identifier of the terminal device is checked in the fixed network. A connection is established from the fixed network to a third party for obtaining a confirmation. A second identifier is transmitted from said terminal device to the third party for identification. The fixed network is replied with the confirmation given by the third party if the third party accepts the second identifier. The terminal device is allowed to use the telecommunication services of the fixed network in response to the confirmation from the third party.

[0015] According to a preferred embodiment of the invention, the use of telecommunication services is charged for on the basis of the confirmation transmitted by a third party.

[0016] According to a preferred embodiment of the invention, a third party identifier, such as a telephone number, is transmitted during a connection establishment from said terminal device to said fixed network, on the basis of which identifier a connection is established to said third party.

[0017] According to a preferred embodiment of the invention, a time limit is set for the duration of a telecommunication connection of said terminal device, a timer is activated when establishing the telecommunication connection of said terminal device and when the time measured by the timer exceeds the time limit, the offering of the telecommunication connection to said terminal device is prevented.

[0018] The invention is based on the idea that when a user (subscriber) of a terminal device, preferably a mobile station, does not have a (direct or an indirect) right to access to the fixed network, i.e. the terminal device cannot transmit an acceptable, a so-called first identifier, e.g. the IMSI of the SIM card, to the network, the fixed network, typically the visited network, allows a connection to be established to a third party preferably for user identification. The terminal device transmits a so-called second identifier, e.g. a digital signature of a payment application, to the third party. The third party identifies the user and, if the user is acceptable, informs the fixed network of having checked the user and preferably of ensuring the charging of the user. Thus, the visited network can offer telecommunication services in a manner requested by the terminal device, since the visited network has advantageously received a confirmation that the telecommunication services used by the terminal device will be paid for. The visited network may charge for the services according to charging instructions that are either received from the third party or are determined in advance. The service charging is addressed for example directly to the third party, which takes care of the further charging of the user. The third party can be contacted in order to verify the creditworthiness, for example, even if the visited network identified the terminal device user by means of a valid SIM card, for example.

[0019] According to a preferred embodiment of the invention, a mobile station transmits a third party identifier to the visited network during a connection set-up.

[0020] According to an embodiment, the visited network may require a confirmation from the third party each time the mobile station requests for a new telecommunication connection or only when the user registers in the network for the first time. Then, in accordance with a preferred embodiment of the invention, the visited network gives the mobile station a temporary user identity either for the duration of one telecommunication connection or for a longer time. The connection established to the third party to identify the user can according to an embodiment be time-supervised, too: if no confirmation from the third party is received within a pre-set time limit, the visited network disconnects the connection.

[0021] The method and apparatus of the invention provide the advantage that when a terminal device having no right to

access to the visited network requests for telecommunication services from the visited network, a confirmation from a third party, e.g. from another mobile communication network or a credit company, can be transmitted to the visited network. On the basis of the confirmation, the visited network can thus have more confidence in receiving the payment for the services, for example, or in the user's identity. The solution of the invention enables a more flexible use of telecommunication services without an agreement made in advance, and the used telecommunication services can preferably be paid for by a current payment method of the third party.

[0022] The solution of the invention may also protect the real identity of a user from the visited network offering telecommunication services. This may be of use for example in a situation, in which a user is offered a chance of voting with his mobile station, whereby only an authority acting as a third party would identify the user. Further, the visited network can receive extra income by charging via the third party also those users for the services that have not made an agreement with the visited network. In accordance with an embodiment of the invention, the visited network can restrict users having no right to access from using telecommunication services preferably by setting a certain time when the offering of telecommunication services is allowed and thus making the risk of misuses smaller.

[0023] In the following the invention will be described in greater detail in connection with the preferred embodiments with reference to the attached drawings, in which

[0024] FIG. 1 shows a simplified view of a mobile communication system in which the invention can preferably be applied;

[0025] FIG. 2 shows a simplified block diagram of an internal structure of a smart card;

[0026] FIG. 3 shows a simplified signalling diagram illustrating the method of the invention;

[0027] FIG. 4 shows a method of encryption and identification according to a public-key technique by way of example.

[0028] The invention can be applied to any telecommunication system in various situations, particularly to mobile communication systems and situations in which the mobile station user establishing a connection with a mobile communication network has no right to use network services. In the solution of the invention, connections can also be established from a mobile communication system to any telecommunication network. In the following, the invention is described in the GSM system according to FIG. 1, the system comprising at least one mobile station MS and a mobile communication network VN, which is a visited network.

[0029] A visited network VN comprises one or more base transceiver stations BTS, which use radio frequencies and channels that are controlled by a base station controller BSC. There is a connection from the base station controller BSC to a mobile services switching center MSC, which is responsible for call set-ups and for routing calls to right addresses. Two databases comprising information on mobile station subscribers are used in this: a home location register HLR and a visitor location register VLR. The home location register HLR comprises information on all subscribers of the visited network VN and the services they have subscribed to, the visitor location register for its part comprises information on mobile stations visiting the mobile services switching center MSC area of the visited network VN. The mobile services switching center MSC and the visitor location register VLR are typically integrated into each other, and the abbreviation MSC/VLR can also be used for the mobile services switching center implementing the functions of the visitor location register VLR.

[0030] The mobile services switching center MSC is in connection with other telecommunication networks, e.g. the PSTN (Public Switched Telephone Network) or the ISDN (Integrated Services Digital Network), via an interworking function IWF. The interworking function IWF is responsible for adjusting the telecommunication between the GSM system and other telecommunication networks. For example, the interworking function IWF typically comprises a modem for the PSTN. A third party 3P may establish a connection to the PSTN. A smart card SC can be inserted into a mobile station MS.

[0031] The description of the invention includes an example, in which a mobile station MS does not have a right to access to the visited network VN and there is no functional SIM card in the mobile station MS. Thus, there is no user identity that can be accepted by the visited network available. The solution of the invention can also be applied to cases, in which the home network providing a SIM card connected to a mobile station MS has no valid roaming agreement with the visited network. The GSM mobile station does not have a right to access to the GSM network either, if the IMEI identifier is on a so-called black list, which means that the mobile station is typically stolen. Mobile stations on the black list, having no right to access to the network on the basis of the IMEI, are not provided with telecommunication services.

[0032] The mobile station MS of the invention can be used without a SIM card, or some other, preferably a smart card SC which is shaped like a SIM card, can be inserted into a SIM card reader or a corresponding card reader. In the following example, a smart card SC comprising a payment application is inserted into a mobile station MS. The payment application of the example is for credit card payments, i.e. the payment application provider charges the party to be charged in arrears. The invention can also be applied to mobile stations MS without a smart card or a payment application. The mobile station MS itself can also comprise a payment application, instead of a smart card SC.

[0033] According to the invention, mobile station MS functions can be used, although a SIM card is not inserted into the mobile station MS. The user inserts a smart card SC comprising a payment application into the mobile station MS. FIG. 2 shows a simplified block diagram 20 of an internal structure of the smart card SC known per se. A smart card SC is typically a plastic card having a size of a credit card, and typically an integrated circuit is laid therein. In addition, there are electric contacts on the smart card SC surface, which help to transmit operating voltages to the card and to transfer control signals and data signals between a smart card reader, such as the mobile station MS and a bus adapter DATA I/O of the smart card SC. Thus, data is transmitted between the smart card SC and the mobile station MS through the bus adapter DATA I/O.

[0034] A central processing unit CPU controls the operation of the smart card on the basis of a program code stored in the read only memory ROM. Different user-specific data, which remain permanently in the memory, can be stored in the electrically erasable programmable read-only memory EEPROM. For example, a payment application may be stored in the EEPROM. The information contained in the smart card SC is arranged in different directories, and the card and external apparatuses have different rights to access to them. Random Access Memory RAM can be used as a temporary storage of information. To ensure a secure usage of the smart card SC has a security function SEC, which checks the PIN, for example. The mobile station MS comprises means for using the smart card SC, e.g. means for reading the electric contacts of the smart card SC and means for writing into the smart card SC memory.

[0035] In the following, the activation of a smart card SC and a payment application contained therein is described by way of example. The smart card SC is inserted into a mobile station MS, which connects an operating voltage to it. The smart card SC transmits the mobile station MS information on its properties, e.g. the protocols it supports and manufacturing information. The mobile station MS is able to use the smart card SC, and the PIN of the mobile station MS user is checked by means of a user interface, e.g. a keyboard, microphone or a pressable display screen. The security function SEC checks, whether the entered PIN is right, after which the smart card SC can be used. This way is can be ensured that only the user knowing the PIN can utilize the smart card SC. A user can also be identified in any other way, for example by using fingerprint recognition. The mobile station MS preferably reads the identifiers of the smart card SC directories, on the basis of which it detects that the smart card in question does not have a SIM card identifier of the GSM. In accordance with the invention, the mobile station MS can, however, be used, and a call to the visited network VN can be set up also when it deals with other services than an emergency call service.

[0036] The mobile station MS can preferably transmit the user the information on applications the smart card SC comprises and on the fact that it does not deal with a SIM card. The user selects a payment application, and the payment application of the smart card SC is activated. A condition for the payment application activation can be a separate user check, for example a separate check of a second PIN2.

[0037] In the following the solution of the invention is illustrated by a simplified signalling diagram of FIG. 3, which shows a mobile MS-originated telecommunication service, whereby no connection is yet established to the visited network VN. In the example of FIG. 3, the invention is applied in the GSM system signalling, but the invention can also be applied in any other telecommunication system signalling. The mobile station MS is activated by switching it on and the smart card SC comprising also a payment application APP is inserted into the mobile station MS and activated for example in the above manner.

[0038] When the mobile station MS user has for example selected the number to be dialled and activated a service request for example by pressing a "call" key, the mobile station begins signalling with the visited network VN. Depending on the implementation, other ways of selecting telecommunication services and detecting the target of the desired connection can also be used. It is also possible that multiple telecommunication services are requested from the visited network VN. However, only one telecommunication service can typically be requested at a time. First, the mobile station MS requests for a free signalling connection (channel request, arrow 30) from the visited network VN. If a free signalling channel is found, the visited network VN transmits the information on the signalling channel to the mobile station MS (immediate assignment, arrow 31).

[0039] Next, the mobile station MS transmits a request for connection origination (service request, arrow 32) on an assigned signalling channel. In a typical GSM call, the mobile station transmits either a TMSI or an IMSI identifier to the visited network in the request for connection origination. The network typically requests for the IMSI separately, if the TMSI is not known. Since neither of the identifiers is available, the mobile station MS either sends a message with an empty field in the space reserved for the TMSI or IMSI or it reports that it does not have a user identity. This can preferably be reported in the same message and in the same form as the IMSI or TMSI, differing, however, from the values reserved for them. The form described for example in the publication WO 9834430 can be used. The publication WO 9834430 describes a data field, in which the bits [000] refer to the type of username that is used for a mobile station having no right to access.

[0040] A third party 3P identifier can also be transmitted in the field reserved for the IMSI/TMSI. In this case, the visited network VN would distinguish on the basis of a certain number, for example, that it deals with a third party identifier and that there is no IMSI identifying the user available. The third party can be contacted in the manner described later. Thus, the visited network VN can already at an early stage of a call set-up check the third party identifier for example from the register of the network. Based on this, the visited network VN can decide, whether it wants to reserve a temporary user identity for the mobile station MS which has no right to access.

[0041] The visited network VN receives the request for connection origination of the mobile station MS, which is preferably interpreted in the mobile services switching center MSCNLR. Then, after detecting that the mobile station MS has not transmitted the user identity, the visited network VN may give a right to a temporary telecommunication connection for the mobile station MS, although it does not contain a SIM card providing the right to access.

[0042] The visited network VN may preferably give the mobile station MS a temporary user identity for example in the manner described in the publication WO 9834430. The visited network VN, preferably the visitor location register VLR, comprises numbers for the temporary user identity and means for marking the numbers as reserved or available. The visited network VN reserves for the mobile station a temporary user identity that is available at that time and transmits it to the mobile station MS in a message, in which a request for check of rights (authentication request, arrow 33) is typically transmitted. Thereafter, the temporary user identity is used in a connection establishment, and when the mobile station MS has received it, it transmits a confirmation message (authentication response, arrow 34). In a typical GSM connection, the authentication of the previous mes-

sages 33 and 34 cannot be performed, since neither a SIM card nor an IMSI is available. The visited network VN preferably uses the assigned temporary user identity as the mobile station MS identifier so long the mobile station MS is in any kind of connection with the visited network VN.

[0043] According to a preferred embodiment of the invention, the visited network VN sets a certain time limit, and when this time limit is exceeded, the telecommunication connection is no longer offered for the mobile station MS. As the visited network VN is not certain of the payment of the use of the telecommunication connections, it may restrict the duration of the mobile station's MS use of the telecommunication connection with the time limit. The visited network VN preferably aims at receiving a confirmation of the payment within the time limit. When the visited network VN has received the confirmation of the payment, the time limit for the use of the telecommunication connection can be deleted.

[0044] The visited network VN comprises a time-measuring timer, preferably in connection with the mobile services switching center MSCNLR. A time limit is stored in connection with the timer. For example, the timer can be switched on, when, after receiving a request for connection origination (service request, arrow 32), the mobile services switching center MSCNLR of the visited network VN detects that the mobile station MS has no right to access to the network, whereby the visited network VN gives the mobile station MS a temporary user identity. The timer compares the time spent on the use of the telecommunication connection assigned to the mobile station with the set time limit. The visited network VN disconnects the telecommunication connection of the mobile station MS, when the timer indicates that the duration of the telecommunication connection has exceeded the determined time limit of the timer.

[0045] Thus, the visited network VN can easily control contacts without access rights and preferably allow a contact with the third party 3P only. If necessary, the visited network VN may preferably change the time limit. For example, if the network is heavily loaded, the time limit may be shorter than in the case the network had a lot of capacity. Other methods exist as well, e.g. a register of the amount of connection attempts can be formed for the mobile station MS on the basis of the IMEI. Thus, the amount of connection attempts can be restricted to the level selected by the visited network VN.

[0046] The visited network preferably transmits a check request of the mobile station equipment identity (IMEI) to the mobile station MS (MS ID request, arrow 35). The mobile station MS transmits the IMEI (MS identity, arrow 36) to the visited network VN. The visited network VN checks the equipment identity register EIR comprising information on the mobile stations for which no telecommunication services are offered. In a GSM call establishment, the mobile services switching center MSCNLR typically transmits the mobile station MS a request for starting an encryption, whereafter the mobile station MS replies to it and starts the encryption. Depending on the mobile station MS implementation, the encryption cannot necessarily be implemented in the solution of the invention, if the mobile station lacks a SIM card, and the encryption activation message need not be transmitted.

[0047] Data that are transmitted over the radio path need not necessarily be encrypted; the payment application of the smart card SC preferably encrypts the information used for identification in a manner which is described later. User data transferred over the radio path need not often be encrypted. The mobile station MS can also comprise an encryption key required for the encryption. For example, a mobile-specific encryption key, which the network can preferably detect by means of the IMEI, may be stored in the mobile station. It is also possible that the visited network transmits the encryption key to be used safely to the mobile station MS during the connection establishment.

[0048] The mobile station MS transmits a connection set-up message (set-up, arrow 37) to the visited network VN. The connection set-up message comprises e.g. the dialled number that the user wants to call and the information on the telecommunication services that are required by the mobile station MS.

[0049] Typically, the GSM network would now, at the stage of having received the target identifier, start to route the connection to the telecommunication network according to the number to be dialled. According to a preferred embodiment of the invention, the visited network VN contacts the third party 3P, which gives the visited network VN a confirmation of paid telecommunication services or the user identity, for example. For this purpose, the mobile station MS preferably transmits the third party 3P identifier to the visited network VN. The third party identifier can already be transmitted when the request for connection origination (service request, arrow 32) has been sent.

[0050] According to a preferred embodiment of the invention, a connection set-up message (set-up, arrow 37) also comprises a third party 3P identifier, preferably a telephone number. The identifier has been received for example from the payment application APP of the smart card SC. The mobile station MS combines the identifier received from the payment application APP with the connection set-up message (set-up, arrow 37) such that it can be distinguished from the actual telephone number of the target to be called. A third party identifier can e.g. be found in the field of a number to be called in the connection set-up message and the telephone number of the target to be called can be transmitted as a so-called Facility message. A Facility message is typically used for transmitting connection-related information between the mobile station and the network either as a separate message or as combined with some other message. The telephone number of the target to be called can preferably be transmitted in the connection set-up message as a Facility message.

[0051] A third party 3P identifier can also be received from the mobile station MS or the user, and it may also be some other type of identifier than a telephone number, an IP address, for instance. The visited network VN detects that the connection set-up message (set-up, arrow 37) comprises a third party 3P identifier and a Facility message. The visited network VN preferably stores the number of the target to be called contained in the Facility message in the memory of the mobile services switching center MSC/VLR, for example.

[0052] The visited network VN starts taking measures in order to establish a connection with the third party 3P on the basis of its identifier. The visited network VN may be aware

of the identifier in advance or the mobile station MS may transmit it in a connection set-up message (set-up, arrow 37) during a call establishment in accordance with the above description. In the current example, a connection to the PSTN network (connection request, arrow 38) is set up on the basis of the third party 3P telephone number. The connection establishment from the GSM network to the PSTN network according to the given telephone number is known from the prior art, and it is not needed to explain herein in greater detail. The connection between the networks can for example be established in accordance with the SS7 standard (Signalling System No. 7). It is obvious that the solution of the invention can also be utilized in establishing a connection with other telecommunication networks, such as the ISDN or PDN (Packet Data Network). The visited network VN and the third party 3P may know one another in advance, and the visited network VN can preferably transmit a certain message, on the basis of which the third party 3P detects that it deals with a confirmation message.

[0053] According to the invention, a third party 3P can be any party comprising means for establishing a telecommunication connection with a visited network VN and for identifying a user or an application used. A third party 3P can be a bank server, for example, whose payment application APP is included in the smart card of the user. Further, a third party 3P may be a server controlled by a public authority, which is capable of identifying the user reliably. A telecommunication system belonging to some other mobile operator can also be a third party 3P, and calls can be made preferably according to the invention without a valid roaming agreement.

[0054] The third party 3P receives a request for connection origination from the visited network VN and a telecommunication connection is established between them according to the prior art (negotiation, arrow 39). Preferably the visited network VN and the third party 3P identify one another, for example with the help of a public-key technique, which will be described later. When a reliable telecommunication connection is established, the visited network VN reserves a traffic channel for the mobile station MS and transmits the information to the mobile station MS (connection open, arrow 40). A data service according to the GSM standard, e.g. an NT (Non-Transparent) connection with the rate of 9600 bits/s, is preferably reserved for the mobile station MS. After this, the telecommunication connection between the mobile station MS and the third party 3P can be used. The mobile station MS further transmits the information to the payment application APP of the smart card (arrow 41).

[0055] In order for the third party 3P to be able to rely on the user, or on the payment application APP of the user, as in the example, and to give a confirmation to the visited network VN, the third party 3P has to identify the payment application APP. The payment application APP transmits its identification data to the mobile station MS, which further transmits the identification data to the visited network VN, which in turn transmits the identification data to the third party 3P (authentication, arrow 42). The identification data preferably verified by a digital signature are transmitted in a safe form such that the visited network VN or the mobile station MS is not able to detect them. It is also possible that the third party 3P requests for the identification data, after which the payment application APP replies by giving the identification data for example in the following way.

[0056] The identification method used varies depending on the used application. One alternative is a so-called public-key cryptography. In the following, a message encryption according to the public-key technique and a digital signature are described in FIG. 4. A digital signature can be used as a preferred embodiment of the invention for authenticating different parties. Typically, when text is encrypted, an algorithm and an encryption key are used, which help to code the unencrypted text to encrypted text. Only the possessor of the right key can correspondingly decode the encrypted text to unencrypted text.

[0057] Two types of keys are used in the public-key technique: public and secret keys. The payment application APP on the smart card SC comprises a secret key Sc, which is only known by the third party 3P and its payment application APP. The third party 3P has the corresponding secret key S3. The payment application APP of the smart card SC further comprises a public key Pc, and the third party has the corresponding key P3. The public-key method is based on the idea that a public key is used in transmitting a message and a secret key in receiving the message. Then anyone who is aware of the public key can code the message, but only the possessor of the right secret key can decode the message.

[0058] In the following, a message encryption and sender identification, in which a digital signature is used and which are implemented according to the public-key method, are described by way of example. A payment application APP comprises two algorithms A and B. A is for encrypting a message and B is for forming a digital signature. A message 50 is delivered to the algorithm B, and by using a secret key Sc, the algorithm B forms a digital signature 51. The message 50 is unencrypted and may comprise information transmitted by the payment application, for example. The digital signature 51 is combined with the message 50 in a summer SUM, and this combination 52 is further delivered to the algorithm A. The algorithm A forms the combination 52 and a message 53 coded by means of a public key Pc. RSA (Rivest-Shamir-Adleman) is a very well-known algorithm used in the public-key method, and it can be used in the solution according to the example.

[0059] The mobile station MS receives the coded message 53 and transmits it further to a visited network VN, which then transmits it to a third party 3P (not shown). Only the third party 3P can decode the encrypted message 53, since it has a secret key S3 relating to the public key Pc of the payment application of the embodiment. The public key Pc can be transmitted in the coded message 53 completely unencrypted, whereby the third party can easily select the right secret key S3. The algorithm A2 of the third party 3P decodes the coded message 53 by using the secret key S3. Compared with the coding algorithm A, the decoding algorithm A2 takes opposite measures. The combined decoded message and digital signature 54 are taken to a separator SEP, which separates the decoded message 55 and the digital signature 56. The decoded message 55 and the digital signature 56 are taken to an algorithm B2, which produces information 57 on the reliability of the digital signature by using the public key P3. On the basis of the information 57 it can be found out, whether the digital signature comes

authentically from the same party as the public key 3P. By using this information, it can thus be ensured that the message sender is the payment application APP on the smart card SC. The algorithm B2 decoding the digital signature is a reversal of the algorithm B.

[0060] This way, the third party 3P can become certain of the reliability of the payment application and of the user reliability as well, since the payment application APP of the smart card SC cannot be activated without the right PIN code. Correspondingly, the public-key method can, if required, be used for identifying the third party 3P in the payment application APP. The public-key method can also be used when the visited network VN and the third party 3P are identifying one another. In addition to the identification, the payment application APP may transmit the third party 3P other information as well, e.g. concerning the properties of the payment application APP.

[0061] Identification data (authentication, arrow 42) transmitted from the mobile station MS to the third party 3P can also be something else than the identifier transmitted by the payment application APP. It may comprise e.g. identification data of the user, the mobile station MS, the smart card SC or some other application. A specific combination of characters, for example, can be used as a user identifier, which is entered by the user by means of a user interface to the mobile station MS.

[0062] The third party 3P receives the identification data (authentication, arrow 42) and identifies the payment application APP reliably and thus the user of the payment application APP. The third party 3P identifies the payment application APP and the user to be its own and is according to a preferred embodiment ready to pay for the telecommunication services to the visited network VN in arrears. The payment may be charged from the user's account on the basis of a previously made agreement on payments.

[0063] The third party 3P preferably transmits a confirmation of the payment of the charge to the visited network VN (confirmation, arrow 43). It is preferable to use a previously agreed practice for confirmation transfer between the third party 3P and the visited network VN. For example, a specific set of numbers, such as 111, is used to indicate that the user in question is reliable and that the payments for the telecommunication connections will be taken care of. Correspondingly, the set of numbers 100 can be used to indicate that the user reliability is not guaranteed and that the third party 3P cannot take care of the payment. Since the visited network VN and the third party 3P have preferably already identified one another in the connection establishment (negotiation, arrow 39), the visited network VN can rely on the third party 3P.

[0064] The confirmation (arrow 43) preferably comprises charging instructions, on the basis of which network elements responsible for the charging are arranged to charge the telecommunication connection offered to the mobile station MS. For example, the information on the third party 3P is stored in the server responsible for the charging, whereby the charging parameters formed during the telecommunication connection are directed to the third party 3P. As a mobile station MS identifier, a temporary user identity or an IMEI code can preferably be used. On the basis of the telecommunication services used by the mobile station MS, the charging center forms a charge which is sent to the third party 3P e.g. electrically. Preferably the charge also comprises an identifier, e.g. the name of the user, transmitted by the third party 3P, and the third party 3P can easily handle the charge. The visited network VN may transmit the information on the service providence and the made payment to the mobile station MS (services available, arrow 44). At this stage, the visited network VN can give up the timer function, as the offered telecommunication connection can be charged.

[0065] As the payment application APP of the example is similar to a credit card, the third party 3P charges the visited network VN in arrears on the basis of the received charge. If the payment application were charged directly in connection with the service, the visited network VN has to be able to operate in the manner defined by the payment application. The third party 3P can transmit exact instructions concerning the charging from the payment application, or the visited network VN and the payment application can negotiate for the payment protocol to be used. In this case, the visited network VN has to transmit the charging information on the used telecommunication services to the payment application on the smart card, which payment application decreases the amount of money stored in the application in accordance with the charging information, for example.

[0066] The solution of the preferred embodiment of the invention provides the advantage that, after taking the previously described measures, the visited network VN can now offer the mobile station MS the services it requires and will receive a confirmation of the payment from the third party 3P. The confirmation (arrow 43) transmitted from the third party 3P to the visited network VN can naturally also be for some other purpose than for the confirmation of the payment, like for delivering information of the user identity.

[0067] The visited network VN can now start the connection establishment to the network according to the number to be called, which is earlier transmitted by the mobile station MS and stored in the visited network VN. The visited network VN thus transmits a request for connection origination to the network according to the number to be called according to the known GSM technique, and the connection is established between the network and the mobile station MS. The telecommunication connection between the visited network VN and the third party 3P can be disconnected and a traffic channel allocated to the mobile station MS can be used, where possible, according to the service required by the mobile station MS. If the traffic channel allocated for identifying the payment application APP is not suitable for the service the mobile station MS requires, the visited network VN preferably allocates the traffic channel according to the request of the mobile station MS and disconnects the connection allocated for the identification of the used payment application APP. Thus, the data connection reserved for the mobile station MS may be disconnected and a traffic channel suitable for a speech connection may be reserved to replace it. If a connection can be established with the dialled telephone number, the mobile station MS user obtains the desired service and the visited network VN can preferably charge the third party 3P for the used service.

[0068] Instead of the IMSI or TMSI of a typical GSM connection, a temporary user identity can preferably be used as user identifier, when the mobile station MS is in connection with the visited network VN. The mobile station MS can

also require new telecommunication connections, to activate the GPRS (General Packet Radio Service) connection, for instance, or to use short message services (SMS). The visited network VN may also demand a confirmation from the third party 3P every time the mobile station requires a new telecommunication connection. It is also possible to restrict the use of the temporary user identity e.g. temporally, by setting a certain time limit for the validity of the user identity, after the third party 3P confirmation has been received. If the time limit of the validity is exceeded, telecommunication services are no longer offered to the mobile station MS. The use of a temporary user identity may be controlled by a timer, and after the timer goes off, a new confirmation must be obtained from the third party.

[0069] If the mobile station MS moves outside of the area of the mobile services switching center of the visited network VN, the connection can be subordinated to another mobile services switching center. Then the temporary user identity used by the mobile services switching center of the visited network VN becomes available and the other mobile services switching center reserves a new temporary user identity for the connection.

[0070] When there are no longer telecommunication connections from the mobile station MS to the visited network VN, the visited network VN can also make the temporary user identity available for the mobile stations which have no right to access and which require telecommunication services. Information on the third party 3P and the user stored in the visited network VN can also be deleted after a charge has been transmitted and paid for. However, if the information is not deleted, it can possibly be used later, if the same payment application is being used and the same mobile station MS without a right to access tries to contact the visited network VN again. Thus, it is still necessary for the third party 3P to identify the user.

[0071] In one geographical area there are often mobile communication networks that are maintained by more than one mobile operator. The mobile station MS can thus search the mobile communication networks that can be distinguished in the coverage area of the mobile station MS and it can inform the user of them on the mobile station MS display, for example. The user can preferably select the mobile communication network of the desired operator, with which network a connection can be established.

[0072] A connection establishment and service providence described above show only one example, but the invention can be applied to other telecommunication networks as well, and the connection can also be established in a different manner than above.

[0073] The invention can also be implemented such that when the mobile station MS without a right to access is activated, i.e. when it is switched on, a connection is established with the visited network VN. Then the mobile station MS is activated to a standby mode in the visited network VN, and the user may activate telecommunication services later. In this case, similar procedures as above can be carried out, and no telephone number to be called nor the information on the telecommunication services required by the mobile station MS are transmitted in a connection set-up message (set-up, arrow 37). Identification data transmitted by the mobile station MS can be checked by the third party 3P in the above manner. On the basis of a confirmation obtained from the third party 3P, the visited network VN can then register the mobile station MS to a standby mode in the same manner as other mobile stations located on its network area, too. If the user requires e.g. a speech service later on, the visited network VN can preferably utilize the temporary user identity assigned already earlier and the information received from the third party 3P, and transmit a request for connection origination directly to the desired telephone number after receiving the connection set-up message.

[0074] According to a preferred embodiment of the invention, a third party 3P can be some wireless telecommunication system. It can be a GSM system, for example; the mobile station MS contacting a visited network VN may have a SIM card, which is provided by a home network that has no valid roaming agreement with the visited network VN. In this case, the visited network VN transmits the identification data, e.g. the IMSI, included in the SIM card to the home network, which identifies the identification data from the home location register HLR and can perform the SIM card authentication. If the SIM card belongs to the home network, the home network preferably gives the confirmation and the required charging instructions to the visited network VN.

[0075] This way the visited network VN could offer telecommunication connections to users whose home network has not made a roaming agreement with a visited network VN earlier. This requires functional telecommunication connections between the visited network VN and the home network. The connection establishment practice can preferably be similar as between the networks that have made a roaming agreement. As the mobile station MS is provided with a SIM card, the traffic over the radio path can be encrypted in a similar manner as in roaming calls according to the prior art. The invention can also be applied when the network operator responsible for the telecommunication network is not the same as the service provider selling telecommunication services. The service provider can in this case be a third party, from which the network operator may require a confirmation.

[0076] The invention is also applicable to mobile-terminated calls. Thus, a so-called user or mobile station call identity would be stored in the visited network, on the basis of which call identity the network could send a page message to a right network in which the mobile station, preferably identified by the third party, is registered. This way it is possible to establish a connection to the mobile station using said call identity.

[0077] According to a preferred embodiment of the invention, a third party can be contacted, even though the visited network identified the mobile station user by means of a valid SIM card, for example. A third party may in this case be e.g. a credit company, which may verify the user's creditworthiness. If the third party gives its confirmation preferably within a certain time limit, services can be offered to the mobile station according to the known GSM technique, for example.

[0078] To make data transmission more effective, a packet radio service GPRS has been developed for the GSM mobile communication networks. The invention is also applicable to the GPRS system (not shown). A mobile station makes a packet-switched attachment (GPRS attach) to the SGSN element (Serving GPRS Support Node) of the GPRS net-

work. If there is no SIM card available, the mobile station cannot transmit the IMSI or TMSI, but it preferably sends a message described above, which does not comprise a user identifier. On the basis of this message the GPRS network detects that the mobile station requiring a connection does not have an acceptable SIM card. The mobile station may further transmit the GPRS network a third party identifier, so that the GPRS network may be certain of the real nature of the connection request. The GPRS network preferably switches on the timer, as described above. If the timer exceeds the value determined for it, the GPRS network disconnects the connection.

[0079] The authentication and encryption messages are not transmitted between the network and the mobile station, as there is no SIM card available. If possible, the GPRS network may also request for the IMEI of the mobile station. The GPRS network informs the mobile station of the accepted attachment, which message preferably also comprises a TLLI identifier (Temporary Logical Link Identity) for a later use. The GPRS network may also reject the connection request by means of an attach reject message.

[0080] When the mobile station has registered in the GPRS network, it sends a so-called PDP (Packet Data Protocol) context activation request, which preferably comprises an IP address of a third party. The third party 3P is contacted and the user or the application can be identified in the manner described above, for example. After identifying the user, the third party may transmit the GPRS network a confirmation, on the basis of which the user is preferably charged. The activated PDP context need not necessarily be removed, but it can be used according to the known GPRS technique in the manner required by the user.

[0081] So-called third generation mobile communication systems have been developed all around the world. These third generation mobile communication systems will utilize similar means for user identification as a SIM card. For example, 3GPP (3rd Generation Partnership Project) is standardising a third generation mobile communication system based on the GSM network, which includes e.g. a new radio interface. A GSM core network will be utilized in the system developed by the 3GPP, whereby the connection control and mobility management will resemble one another considerably.

[0082] USIM (Universal Service Identity Module) is an application standardised by the 3GPP, which is based on the SIM card of the GSM system and which will be used for user identification, for example. To implement interoperability, e.g. a handover, between the 3GPP third generation mobile communication system and the GSM system, a GSM SIM functionality along the USIM application may be implemented as such in the same smart card. The invention is also applicable to the third generation mobile communication systems.

[0083] It will be apparent to a person skilled in the art that as the technique develops, the basic idea of the invention may be implemented in a variety of ways. The invention may preferably also be applied to other terminal devices than mobile stations, such as to telephone boxes or a computer with functions required by a mobile station. Thus, the invention and the embodiments thereof are not restricted to the above examples, but may be modified within the scope of the claims.

1. A method of providing telecommunication services in a telecommunication system, the system including at least one terminal device and a fixed network providing network access for said terminal device, the method comprising:

establishing a connection from said terminal device to said fixed network,

checking a first identifier of said terminal device in said fixed network,

establishing a connection from said fixed network to a third party for obtaining a confirmation,

transmitting a second identifier from said terminal device to the third party for identification,

replying to said fixed network with the confirmation given by the third party in response to the third party accepting the second identifier, and

allowing said terminal device to use the telecommunication services of said fixed network in response to the confirmation of the third party.

2. A method as claimed in claim 1, wherein the connection from said fixed network to the third party is established in response to said terminal device not having the first identifier accepted by said fixed network.

3. A method as claimed in claim 1 or 2, further comprising charging for the use of telecommunication services on the basis of the confirmation transmitted by the third party.

4. A method as claimed in claim 3, wherein the third party is charged for the use of the telecommunication services.

5. A method as claimed in any one of the preceding claims, wherein a third party identifier, such as a telephone number, is transmitted during the connection establishment from said terminal device to said fixed network, on the basis of which identifier the connection is established to the third party.

6. A method as claimed in claim 1, further comprising:

setting a time limit for the duration of a telecommunication connection of said terminal device,

activating a timer when establishing the telecommunication connection of said terminal device, and

preventing the offering of the telecommunication connection to said terminal device when the time measured by the timer exceeds the time limit.

7. A method as claimed in claim 1, further comprising reserving a temporary user identity for said terminal device to said fixed network.

8. A method as claimed in claim 7, further comprising:

setting a time limit for the validity of said user identity, and

rejecting to offer telecommunication services for said terminal device if the time limit is exceeded.

9. A method as claimed in claim 1, wherein the third party identifies a terminal device user via the second identifier.

10. A method as claimed in claim 9, wherein the third party identifies the user every time the user requires a telecommunication service.

11. A method as claimed in claim 1, wherein said telecommunication system is wireless and said terminal device is a mobile station.

12. A method as claimed in claim 1, wherein said terminal device is provided with a smart card, whose application includes the second identifier, which is transmitted to the third party.

13. A method as claimed in claim 12, wherein the application of said smart card is a payment application comprising the second identifier.

14. A method as claimed in claim 12, wherein

the application of said smart card is a SIM application, and

that the third party is another wireless telecommunication system.

15. A method as claimed in claim 1, wherein a public-key identification method is employed.

16. A telecommunication system comprising at least one terminal device and a fixed network providing network access for said terminal device, wherein

said fixed network is configured to check a first identifier of said terminal device, said fixed network is configured to establish a connection to a third party for obtaining a confirmation,

said terminal device is configured to transmit a second identifier to the third party for identification,

the third party is configured to transmit the confirmation to said fixed network in response to the third party accepting the second identifier, and

said fixed network is configured to offer telecommunication services for said terminal device in response to the confirmation of the third party.

17. A system as claimed in claim 16, wherein said fixed network is configured to establish a connection to the third party in response to said terminal device not having the first identifier accepted by said fixed network.

18. A system as claimed in claim 16 or 17, wherein said fixed network is configured to charge for the use of telecommunication services on the basis of the confirmation transmitted by the third party.

19. A system as claimed in claim 16, 17 or 18, wherein said terminal device is configured to transmit a third party identifier, such as a telephone number, during a connection establishment to said fixed network, on the basis of which identifier the fixed network is configured to establish a connection to the third party.

20. A system as claimed in claim 16, wherein said fixed network is configured to reserve a temporary user identity for said terminal device to the fixed network.

21. A system as claimed in claim 16, wherein

said telecommunication system is wireless, and

said terminal device is a mobile station.

22. A system as claimed in claim 16, wherein

said terminal device comprises means for using a smart card in the terminal device, the terminal device, and

said fixed network is configured to transmit the second identifier of an application of the smart card to the third party.

23. A system as claimed in claim 22, wherein the application of the smart card is a payment application comprising the second identifier.

24. A system as claimed in claim 22, wherein the application of said smart card is a SIM application and that the third party is another wireless telecommunication system.

25. A system as claimed in claim 16, wherein at least said smart card and the third party are configured to employ a public-key identification method.

26. A network element of a telecommunication system, wherein

said network element is configured to provide network access for at least one terminal device and check a first identifier of said terminal device,

said network element is configured to establish a connection to a third party in response to said terminal device not having the first identifier accepted by said network element, and

said network element is configured to offer telecommunication services for said terminal device in response to a confirmation from the third party.

27. A network element as claimed in claim 26, wherein the network element is a mobile services switching center of a wireless telecommunication system.

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United States Patent [19]

Baker

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[45] **Date of Patent: Jul. 2, 1996**

[54] **TELECOMMUNICATION SYSTEM WITH USER MODIFIABLE PBX TERMINATING CALL FEATURE CONTROLLER AND METHOD**

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5,384,841	1/1995	Adams et al.	379/266

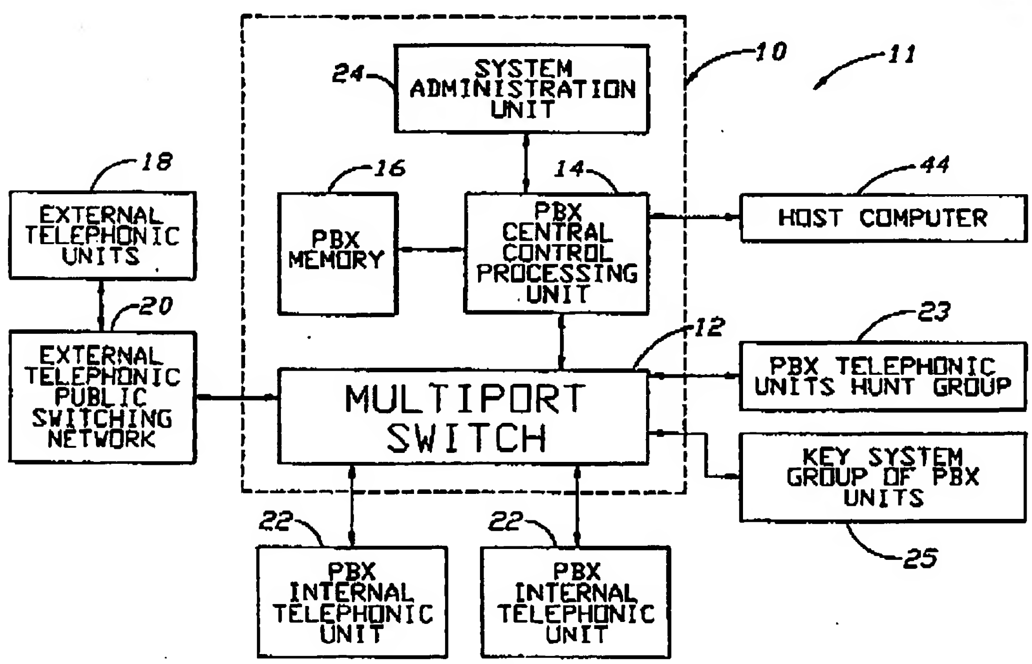
Primary Examiner—Jeffery Hofsass
Assistant Examiner—Scott Wolinsky
Attorney, Agent, or Firm—C. B. Patti; G. A. Montayne

[57] **ABSTRACT**

A telecommunication system (11) having a private branch exchange (PBX) (10) with a multiport switch (12) controlled by a central control processing unit (14) and an associated PBX memory (16) for directing telephonic calls to terminate at identified ones of a plurality of PBX internal telephonic units (22) placed at predetermined positions of the multiport switch (12), a user modifiable PBX terminating call feature controller (30) which stores a call handling feature script in memory (16) defining call handling operations for telephonic calls directed to an identified PBX internal telephonic unit (22) and a system administration unit (24) coupled with the central control processing unit (14) of the PBX (10) for modifying the call handling feature script through employment of commands entered at the system administration unit (24) at the PBX (10) to alter the call handling operations of telephonic calls terminating at the identified PBX internal telephonic unit (22).

18 Claims, 9 Drawing Sheets

COPY



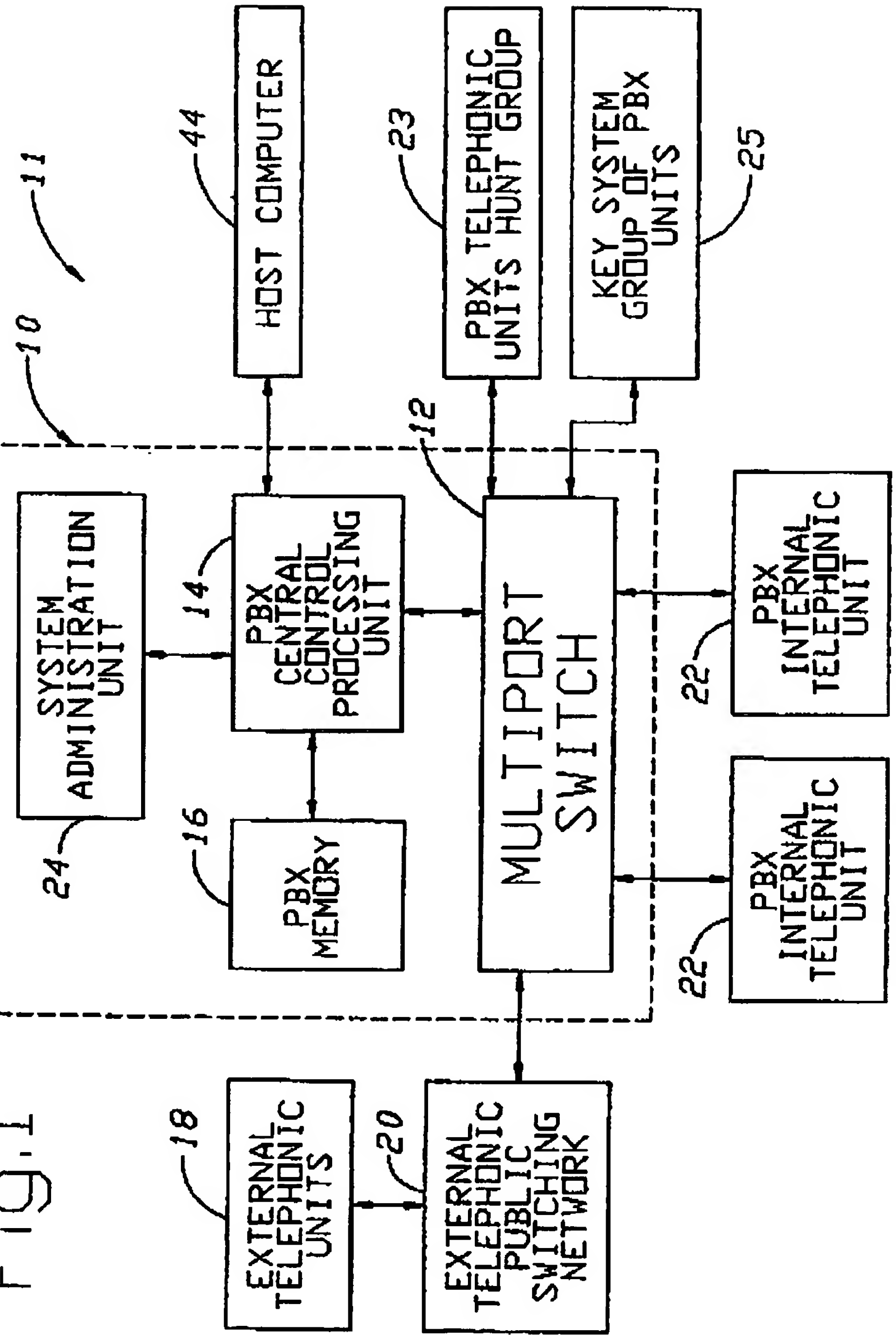

11
10
44
HOST COMPUTER
23
PBX TELEPHONIC UNITS HUNT GROUP
KEY SYSTEM GROUP OF PBX UNITS
25
12
SYSTEM ADMINISTRATION UNIT
14
PBX CENTRAL CONTROL PROCESSING UNIT
MULTIPORT SWITCH
22
PBX INTERNAL TELEPHONIC UNIT
24
16
PBX MEMORY
22
PBX INTERNAL TELEPHONIC UNIT
18
EXTERNAL TELEPHONIC UNITS
20
EXTERNAL TELEPHONIC PUBLIC SWITCHING NETWORK

Fig.1

Fig.2

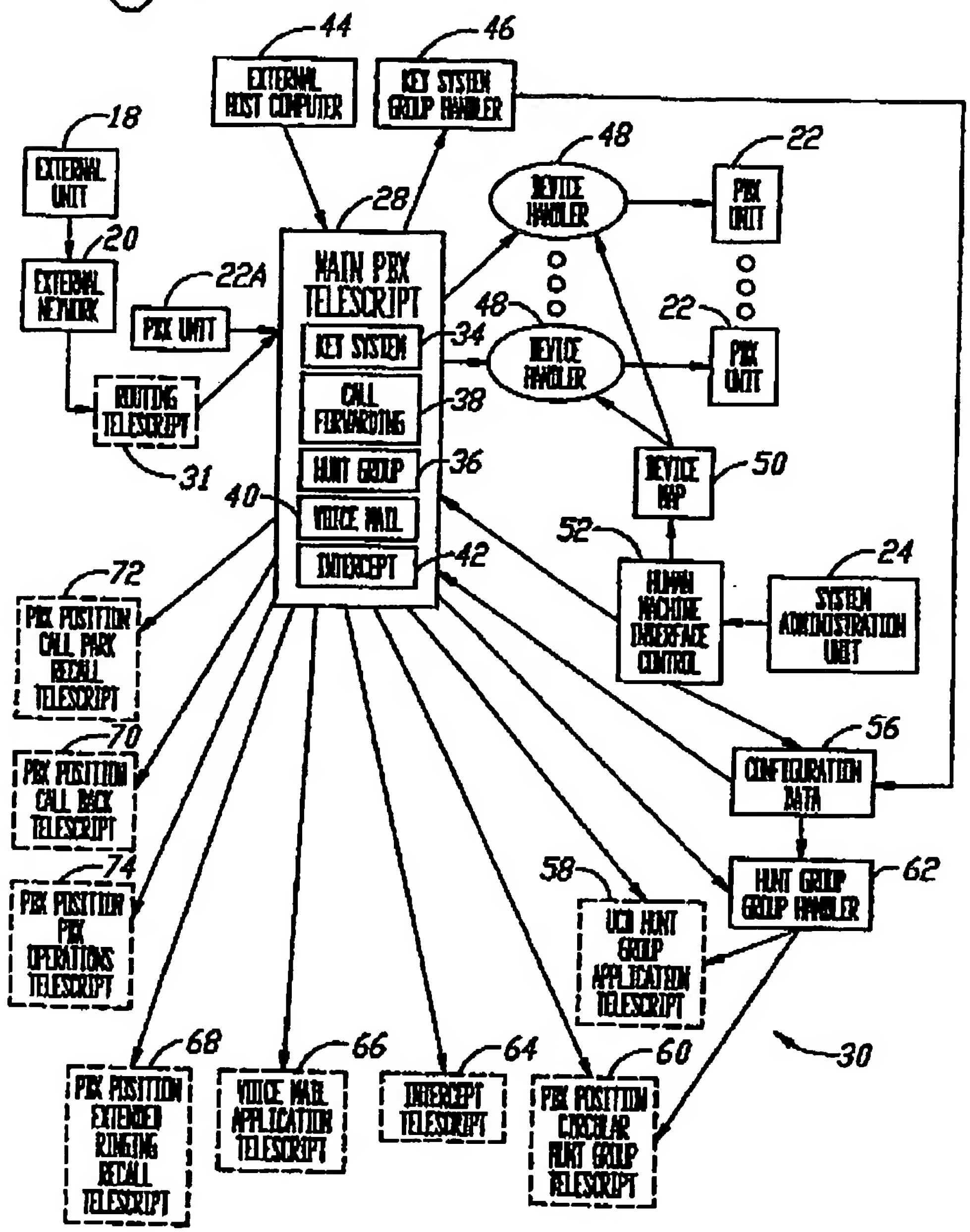
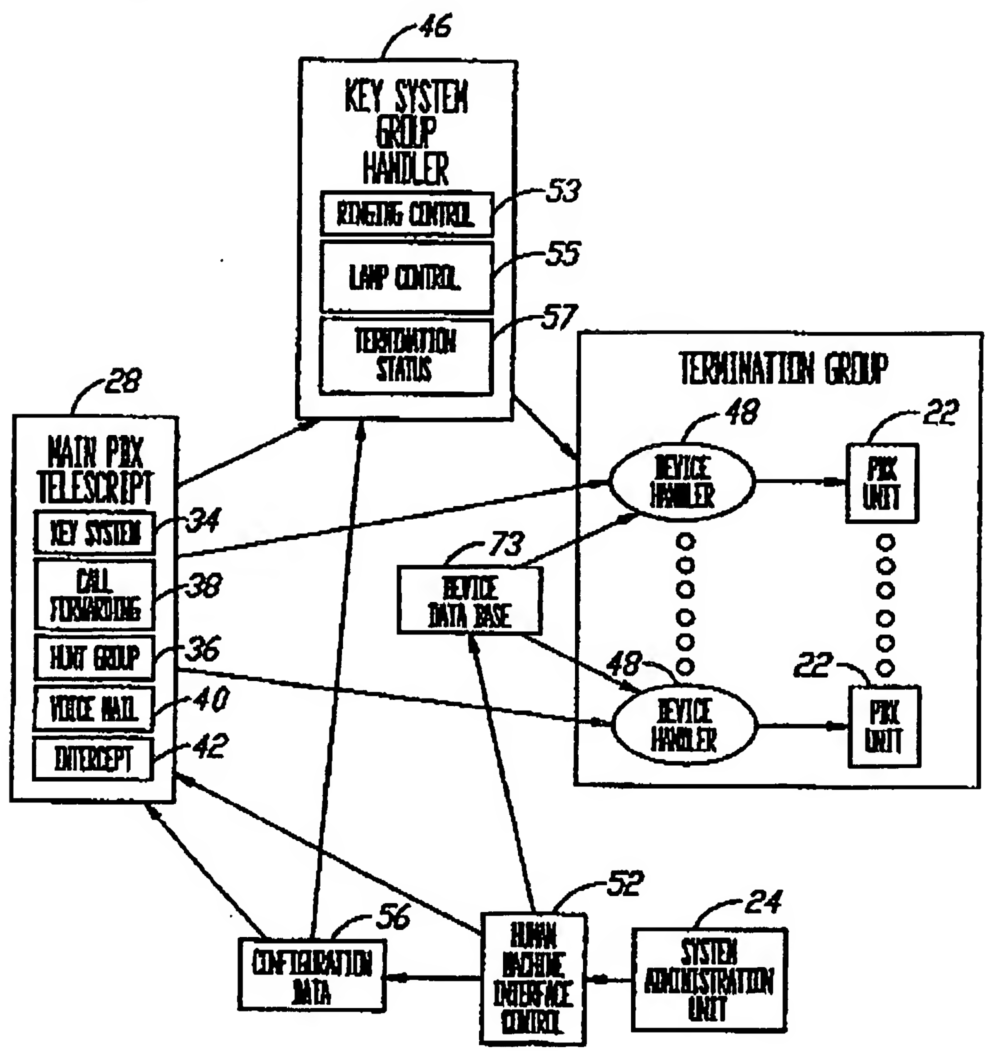

EXTERNAL HOST COMPUTER
44
KEY SYSTEM GROUP HANDLER
46
EXTERNAL UNIT
18
EXTERNAL NETWORK
20
PBX UNIT
22A
ROUTING TELESCRIPT
31
MAIN PBX TELESCRIPT
28
KEY SYSTEM
34
CALL FORWARDING
38
HUNT GROUP
36
VOICE MAIL
40
INTERCEPT
42
DEVICE HANDLER
48
PBX UNIT
22
DEVICE MAP
50
HUMAN MACHINE INTERFACE CONTROL
52
SYSTEM ADMINISTRATION UNIT
24
CONFIGURATION DATA
56
HUNT GROUP GROUP HANDLER
62
UCD HUNT GROUP APPLICATION TELESCRIPT
58
PBX POSITION CALL PARK RECALL TELESCRIPT
72
PBX POSITION CALL BACK TELESCRIPT
70
PBX POSITION PBX OPERATIONS TELESCRIPT
74
PBX POSITION EXTENDED RINGING RECALL TELESCRIPT
68
VOICE MAIL APPLICATION TELESCRIPT
66
INTERCEPT TELESCRIPT
64
PBX POSITION CIRCULAR HUNT GROUP TELESCRIPT
60
30

Fig.3


46
KEY SYSTEM GROUP HANDLER
RINGING CONTROL
53
LAMP CONTROL
55
TERMINATION STATUS
57
TERMINATION GROUP
48
DEVICE HANDLER
22
PBX UNIT
28
MAIN PBX TELESCRIPT
KEY SYSTEM
34
CALL FORWARDING
38
HUNT GROUP
36
VOICE MAIL
40
INTERCEPT
42
73
DEVICE DATA BASE
48
DEVICE HANDLER
22
PBX UNIT
52
HUMAN MACHINE INTERFACE CONTROL
56
CONFIGURATION DATA
24
SYSTEM ADMINISTRATION UNIT

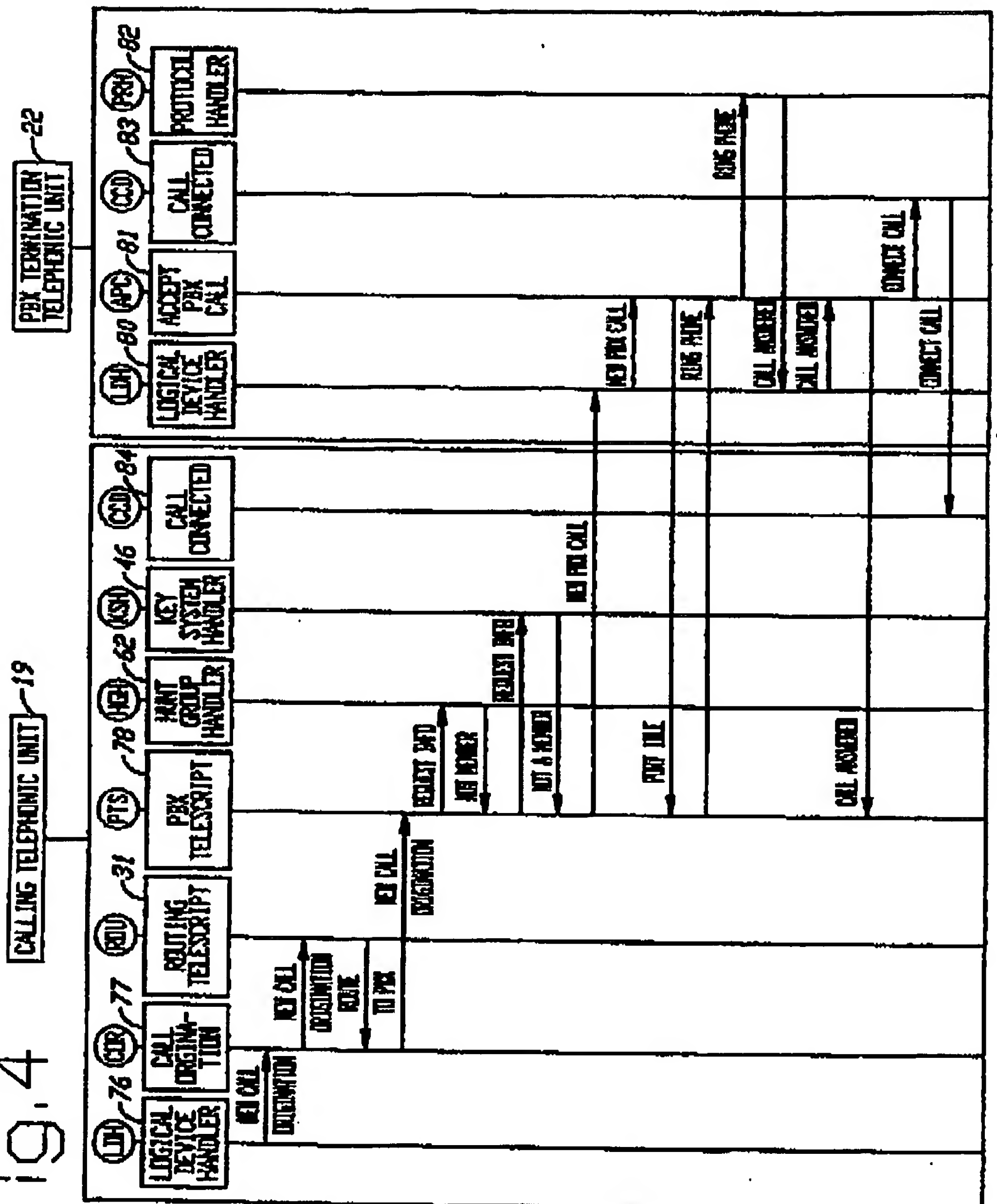
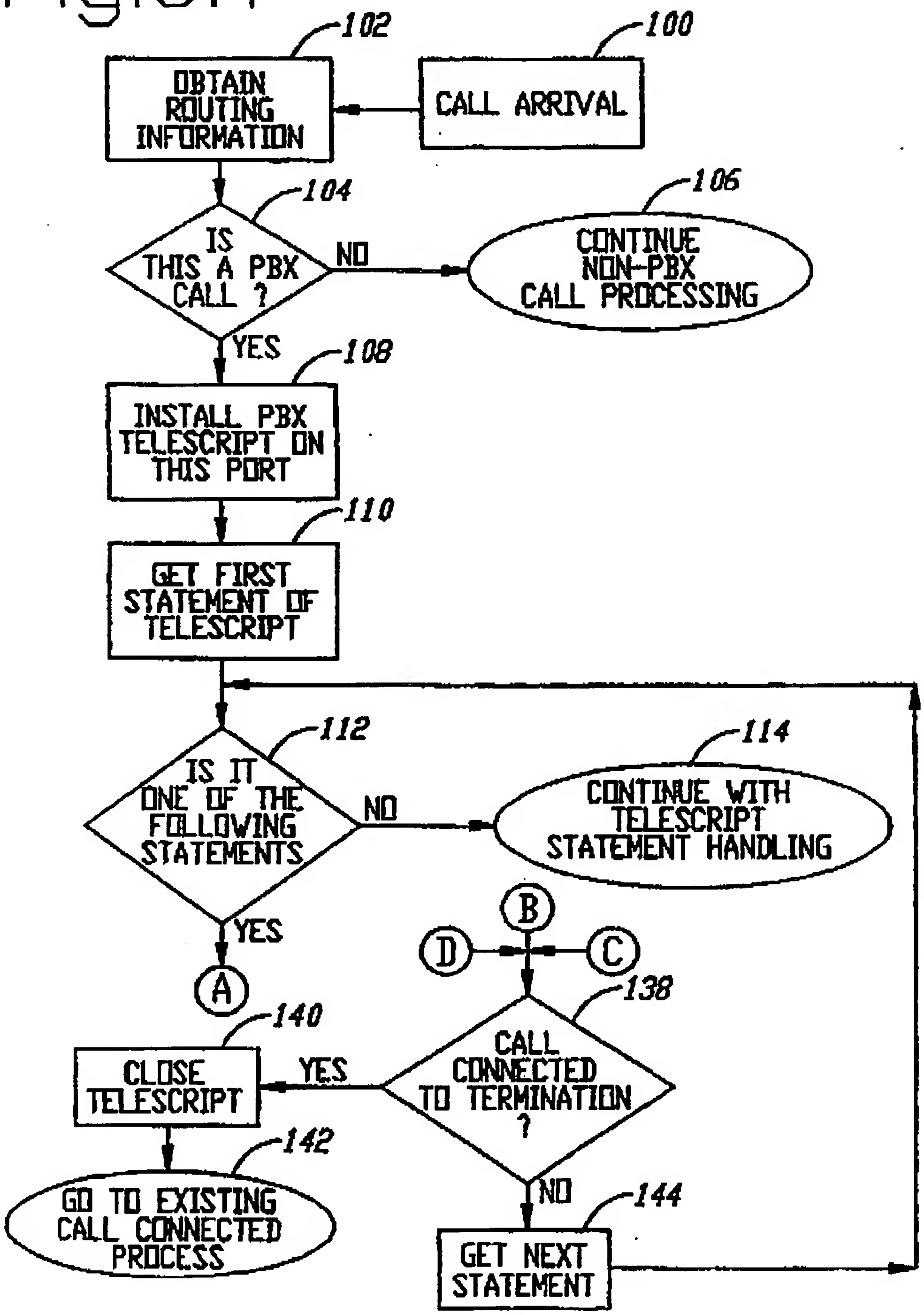

PBX TERMINATION TELEPHONIC UNIT
22
CALLING TELEPHONIC UNIT
19
LDH
76
LOGICAL DEVICE HANDLER
COR
77
CALL ORIGINATION
31
ROUTING TELESCRIPT
PTS
78
PBX TELESCRIPT
HGH
62
HUNT GROUP HANDLER
KSH
46
KEY SYSTEM HANDLER
CCD
84
CALL CONNECTED
LDH
80
LOGICAL DEVICE HANDLER
APC
81
ACCEPT PBX CALL
CCD
83
CALL CONNECTED
PRH
82
PROTOCOL HANDLER
NEW CALL ORIGINATION
NEW CALL ORIGINATION ROUTE
TO PBX
NEW CALL ORIGINATION
REQUEST INFO
ARE MEMBER
REQUEST INFO
NOT A MEMBER
NEW PBX CALL
PORT IDLE
CALL ANSWERED
NEW PBX CALL
RING PHONE
CALL ANSWERED
CALL ANSWERED
RING PHONE
CONNECT CALL
CONNECT CALL

Fig. 4

Fig.5A



102
100
OBTAIN ROUTING INFORMATION
CALL ARRIVAL
104
106
IS THIS A PBX CALL ?
NO
CONTINUE NON-PBX CALL PROCESSING
YES
108
INSTALL PBX TELESCRIPT ON THIS PORT
110
GET FIRST STATEMENT OF TELESCRIPT
112
114
IS IT ONE OF THE FOLLOWING STATEMENTS
NO
CONTINUE WITH TELESCRIPT STATEMENT HANDLING
YES
A
B
D
C
138
140
CLOSE TELESCRIPT
YES
CALL CONNECTED TO TERMINATION ?
142
GO TO EXISTING CALL CONNECTED PROCESS
NO
144
GET NEXT STATEMENT

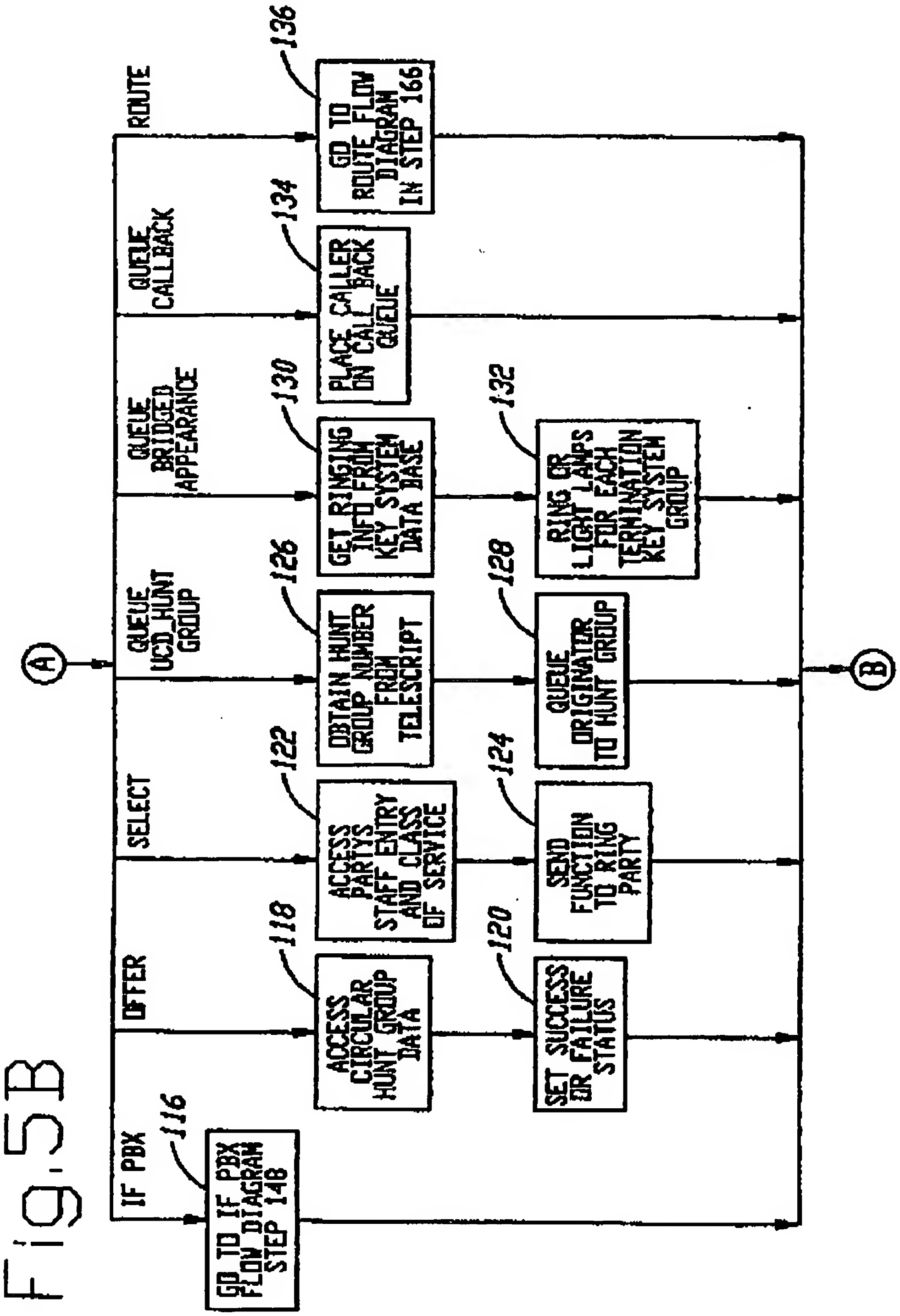
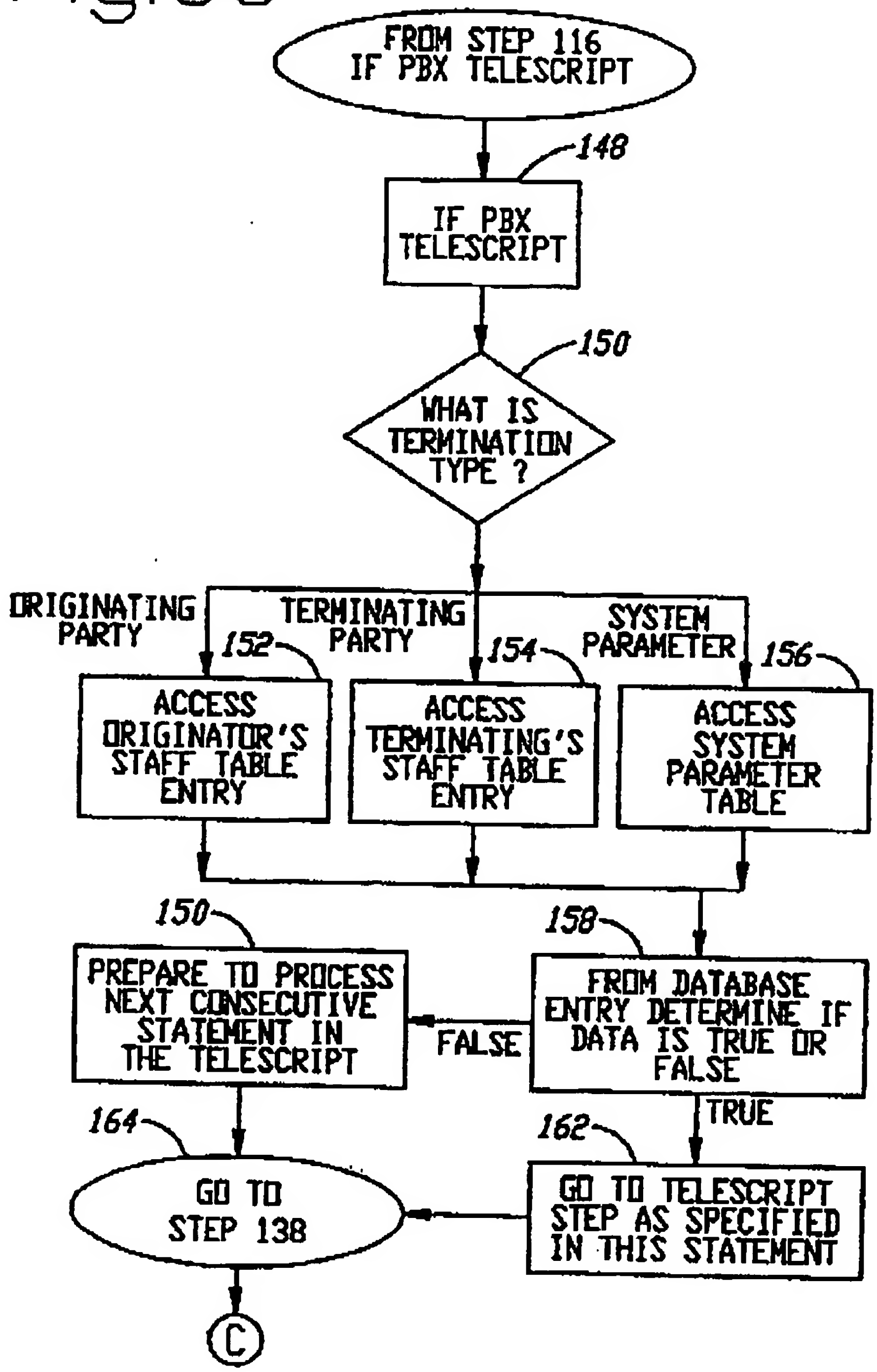
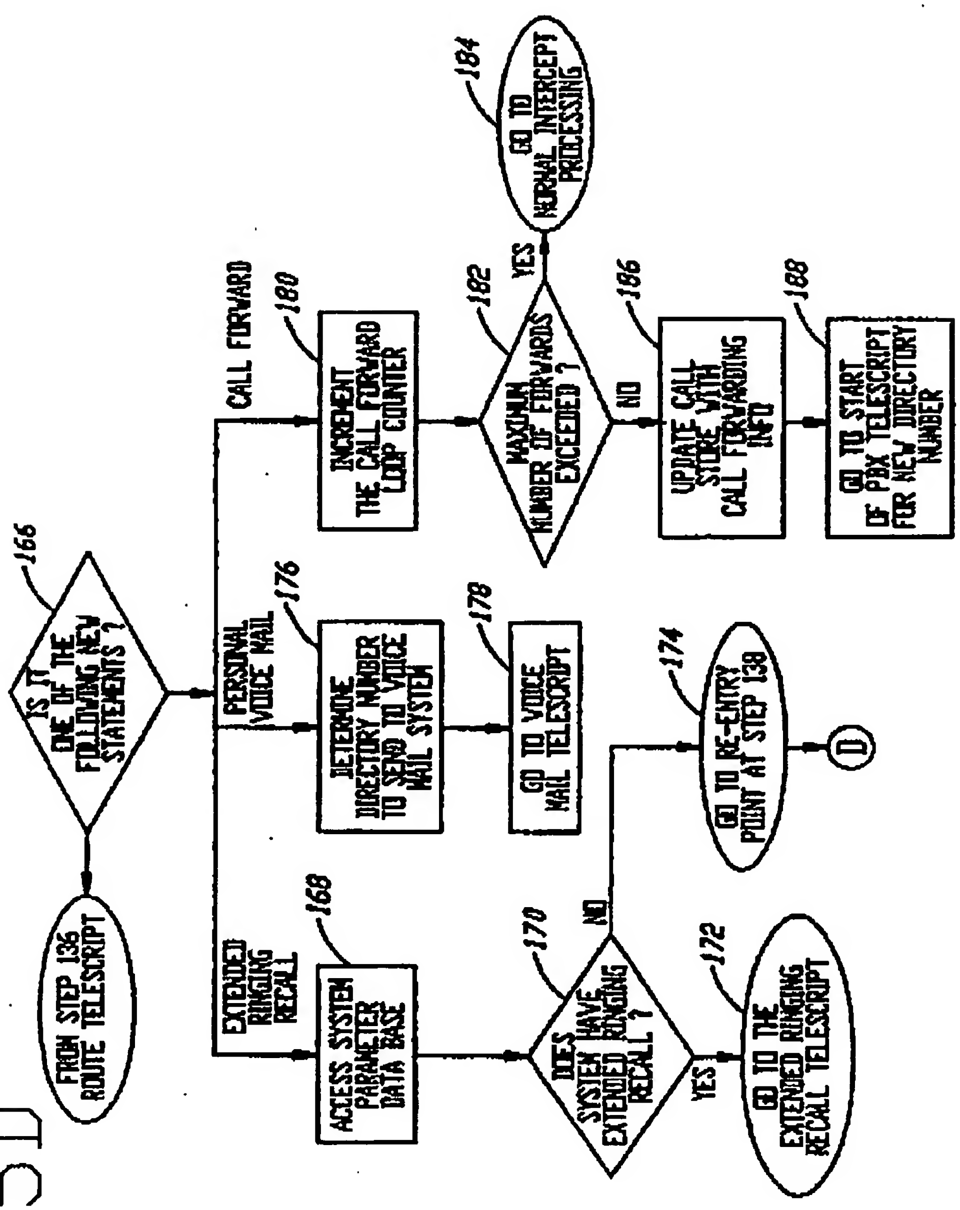

A
B
IF PBX
OFFER
SELECT
QUEUE UCD_HUNT GROUP
QUEUE BRIDGED APPEARANCE
QUEUE CALLBACK
ROUTE
116
GO TO IF PBX FLOW DIAGRAM STEP 148
118
ACCESS CIRCULAR HUNT GROUP DATA
120
SET SUCCESS OR FAILURE STATUS
122
ACCESS PARTYS STAFF ENTRY AND CLASS OF SERVICE
124
SEND FUNCTION TO RING PARTY
126
OBTAIN HUNT GROUP NUMBER FROM TELESCRIPT
128
QUEUE ORIGINATOR TO HUNT GROUP
130
GET RINGING INFO FROM KEY SYSTEM DATA BASE
132
RING OR LIGHT LAMPS FOR EACH TERMINATION KEY SYSTEM GROUP
134
PLACE CALLER ON CALL BACK QUEUE
136
GO TO ROUTE FLOW DIAGRAM IN STEP 166

Fig.5B

Fig.5C



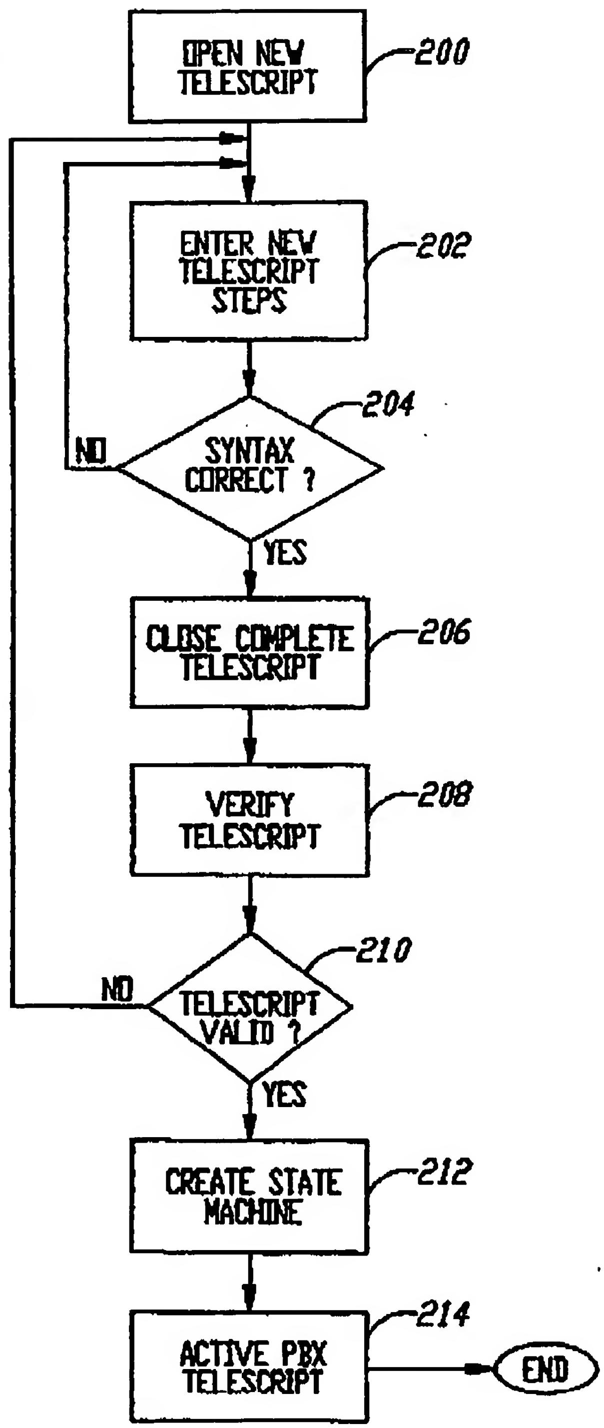
FROM STEP 116 IF PBX TELESCRIPT
148
IF PBX TELESCRIPT
150
WHAT IS TERMINATION TYPE ?
ORIGINATING PARTY
TERMINATING PARTY
SYSTEM PARAMETER
152
154
156
ACCESS ORIGINATOR'S STAFF TABLE ENTRY
ACCESS TERMINATING'S STAFF TABLE ENTRY
ACCESS SYSTEM PARAMETER TABLE
150
158
PREPARE TO PROCESS NEXT CONSECUTIVE STATEMENT IN THE TELESCRIPT
FALSE
FROM DATABASE ENTRY DETERMINE IF DATA IS TRUE OR FALSE
TRUE
164
162
GO TO STEP 138
GO TO TELESCRIPT STEP AS SPECIFIED IN THIS STATEMENT
C



FROM STEP 136 ROUTE TELESCRIPT
166
IS IT ONE OF THE FOLLOWING NEW STATEMENTS ?
EXTENDED RINGING RECALL
PERSONAL VOICE MAIL
CALL FORWARD
168
ACCESS SYSTEM PARAMETER DATA BASE
170
DOES SYSTEM HAVE EXTENDED RINGING RECALL ?
YES
NO
172
GO TO THE EXTENDED RINGING RECALL TELESCRIPT
174
GO TO RE-ENTRY POINT AT STEP 138
D
176
DETERMINE DIRECTORY NUMBER TO SEND TO VOICE MAIL SYSTEM
178
GO TO VOICE MAIL TELESCRIPT
180
INCREMENT THE CALL FORWARD LOOP COUNTER
182
MAXIMUM NUMBER OF FORWARDS EXCEEDED ?
YES
NO
184
GO TO NORMAL INTERCEPT PROCESSING
186
UPDATE CALL STORE WITH CALL FORWARDING INFO
188
GO TO START OF PBX TELESCRIPT FOR NEW DIRECTORY NUMBER

Fig.5D

Fig.6



OPEN NEW TELESCRIPT
200
ENTER NEW TELESCRIPT STEPS
202
SYNTAX CORRECT ?
204
NO
YES
CLOSE COMPLETE TELESCRIPT
206
VERIFY TELESCRIPT
208
TELESCRIPT VALID ?
210
NO
YES
CREATE STATE MACHINE
212
ACTIVE PBX TELESCRIPT
214
END

TELECOMMUNICATION SYSTEM WITH USER MODIFIABLE PBX TERMINATING CALL FEATURE CONTROLLER AND METHOD

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates generally to the field of private branch exchange telecommunication systems and, more particularly, to a private branch exchange system having various call servicing features for calls terminating at private branch exchange telephonic units of the system.

2. Description of the Prior Art

Telecommunication systems having a private branch exchange (PBX) controlled by a central control processing unit in conjunction with a main memory for connecting telephonic calls received from external telephonic units of an external switching network with private branch exchange telephonic units at predetermined positions of the PBX are well known. Examples of such systems are shown in U.S. Pat. No. 5,268,903 of Jones et al. entitled "Multichannel Telephonic Switching Network With Different Signaling Formats and Cross Connect/PBX Treatment Selectable For Each Channel", issued Dec. 7, 1993; U.S. Pat. No. 5,140,611 of Jones et al. entitled "Pulse Width Modulated Self-Clocking and Self-Synchronizing Data Transmission and Method for a Telephonic Communication Network Switching System", issued Aug. 18, 1992; U.S. Pat. No. 5,127,004 of Lenihan et al. entitled "Tone and Announcement Message Code Generator for a Telephonic Switching System and Method" issued Jun. 30, 1992 and U.S. Pat. No. 4,627,047 of Pitroda et al. entitled "Integrated Voice and Data Telecommunication Switching System", issued Dec. 2, 1986.

In PBX systems, telephonic calls are switched to a particular line identified by the dialed call and connected to a PBX telephonic unit associated with the telephone line. Unlike, in automatic call distributor systems in which calls are routed to various groups or pools of agents for servicing, PBX systems are programmed to route a received call directly to the PBX position or PBX telephonic unit identified by the call dialed from the calling party. A call terminating at a PBX position telephonic unit that is busy or is set not to accept calls, is handled by one or more different call servicing features (e.g. call forwarding, subsequent call back, forward to voice mail, etc.) of the PBX system. The feature types and the order of the procedural steps for handling a PBX call directed at an associated PBX position telephonic unit are based on programmed and stored 'C' code in known PBX systems. The order in which the terminating features for a call associated with a particular PBX position are checked by the system against any feature specific decisions about call handling residing in the compiled and stored software code.

The compiled code checks the current state of a call along with provisionable feature parameters to implement decisions on call handling. Therefore, PBX systems operate according to the fixed internal software feature ordering. Disadvantageously, the stored and compiled code is not readily alterable without the necessity of employing a skilled programmer who must accurately reprogram the system to the desired change in feature ordering without interfering with the internal operation of the system.

In certain automatic call distribution systems various call handling features or applications associated with particular agent groups are alterable by the user without reprogramming the compiled code internally stored at the call distributor. This type of alterable call feature handling in an automatic call distributor is seen in U.S. patent application Ser. No. 07/975,240 of Sumner et al. entitled "Control Device For The Network Of A Telephone Switching System" filed Nov. 12, 1992. Unfortunately, the user alterable call feature handling for automatic call distributors is not seen in PBX systems. Moreover, the operation of a PBX system significantly differs from that of an automatic call distributor. Automatic call distributions are very structured for the specific purpose of routing calls to agent groups for servicing a high volume of calls in a short time. Agent consoles in automatic call distributors are generally limited to application features relating to optimizing the agent handling of a call by minimizing the discrete actions required to answer and process a call. In automatic call distribution systems, calls are routed to agent groups with the corresponding agents having very little or no control in the method of routing or of answering received calls. A PBX position telephonic unit, conversely, is a general purpose device having call handling features which enable the called party to either answer a call or send the call to alternate destinations (e.g. voice mail). However, in known PBX systems, a fixed order of operation is set in the compiled code for calls associated with a PBX position. The user or system administrator has no control over the execution of PBX calls since the PBX system operates only according to its internal software feature ordering. Once a particular order of feature operation is determined, in known systems, the set operation is fixed in compiled code software modules internal to the system. Disadvantageously, an operational change in the features for PBX positions, in a known system, requires the extremely time consuming and highly skilled task of reprogramming the internally stored compiled feature application software code. Therefore, even the slightest modification to feature operations are time consuming, inefficient and costly to the user due to the inflexibility of known PBX systems.

SUMMARY OF THE INVENTION

It is therefore a principal object of the present invention to provide a telecommunication system having a private branch exchange (PBX) with a user modifiable PBX terminating call feature controller and method in which the disadvantages of known PBX systems noted above are overcome by providing means and methods for selectively altering the call handling operations of telephonic calls terminating at PBX internal telephonic units.

The object is achieved by providing a telecommunication system having a private branch exchange (PBX) with a multiport switch controlled by a central control processing unit and an associated PBX main memory for directing telephonic calls to terminate at identified ones of a plurality of PBX internal telephonic units placed at predetermined positions of the multiport switch, with a user modifiable terminating call feature controller having means for storing a call handling feature script defining call handling operations for telephonic calls directed to an identified PBX internal telephonic unit and means for modifying the call handling feature script through the employment of commands entered at the PBX to alter the call handling operations of telephonic calls terminating at the identified PBX internal telephonic unit.

The object is achieved by performing a method for modifying PBX terminating call features in a telecommunication system having a private branch exchange (PBX) with a multiport switch controlled by a central control

processing unit and an associated PBX main memory for directing telephonic calls to terminate at identified ones of a plurality of PBX internal telephonic units placed at predetermined positions of the multiport switch comprising the steps of (a) storing a call handling feature script defining call handling operations for telephonic calls terminating at an identified PBX internal telephonic unit and (b) modifying the call handling feature script through the employment of commands entered while the PBX is on-line to alter the call handling operations of telephonic calls terminating at the identified PBX internal telephonic unit.

BRIEF DESCRIPTION OF THE DRAWING

The foregoing objects and advantageous features of the invention will be explained in greater detail and others will be made apparent from the detailed description of the preferred embodiment of the present invention which is given with reference to the several figures of the drawing, in which:

FIG. 1 is a functional block diagram of the preferred embodiment of the telecommunication system with the private branch exchange (PBX) of the present invention as interconnected with a known external telephonic network;

FIG. 2 is a block diagram of the preferred PBX call feature controller present invention for the PBX of FIG. 1;

FIG. 3 is a block diagram of the key system control apparatus of the present invention;

FIG. 4 is a preferred bounce diagram for a call terminating at a PBX internal telephonic unit of FIG. 1 of the present invention;

FIGS. 5A-5D form a composite flow chart illustrating the preferred procedural processing flow for telescript statements of the present invention; and

FIG. 6 is a flow chart for the preferred procedural steps for modifying a telescript of the present invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring to FIG. 1, a private branch exchange (PBX) 10 is shown having a multiport switch 12 controlled by a PBX central processing unit 14 in conjunction with an associated PBX main memory 16 for directly connecting telephonic calls made from external telephonic units 18 via an external telephonic switching network 20 to correspondingly identified PBX internal telephonic units 22. The telephone number dialed at an external telephonic unit 18 identifies a particular individual telephonic line of a PBX internal telephonic unit 22 coupled with the PBX 10. The PBX 10 directly routes an identified telephonic call received at the multiport switch 12 to the associated PBX telephonic unit 22.

A system administrator stationed at a system administration unit 24 coupled with the PBX central control processing unit 14 enters various high level commands at the administration unit to modify procedural order of execution and the types of call servicing features implemented at the PBX internal telephonic units 22. The system administration unit 24 is preferably either a terminal or personal computer which interfaces with the PBX central processing unit 14 to alter and modify the various telescripts stored in the PBX main memory 16.

A telescript is a set or series of call handling coded statements which are associated with PBX telephonic units 22. A defined telescript determines the type, order and priority of call servicing features for a particular terminating position or PBX telephonic unit 22. For example, if a PBX call is routed to a corresponding PBX position telephonic unit 22 and the unit is busy or set not to receive a call the defined PBX telescript associated with the PBX unit determines how the call is serviced within the system (i.e. call routed to voice mail, music played to external unit, route to alternate PBX position etc.). The call servicing features are readily modified by the user or system administrator through implementation of various telescript commands at the system administration unit 24. Any modification to the ordered high level statements comprising a new telescript are verified by the PBX central control processing unit 14 to ensure the modifications comply with the internal call servicing parameters of the PBX 10 of the telecommunication system 11.

A call handling feature script, called a telescript, is associated with each port of the multiport switch 12 which is identified as being connected to a PBX internal telephonic unit 22. The call handling feature scripts define the call handling operations for dialed telephonic calls identified to terminate at PBX internal telephonic units 22. The call handling feature script or telescript is an ordered set of call handling statements with each statement defining a particular call servicing or switching operation. The relatively high level ordered statements are composed of relatively low level coded instructions which run on the PBX central control processing unit 14 to execute the call handling operations for calls terminating at PBX internal units 22 of the switch 12. The call handling feature script consisting of the relatively high level ordered call handling statements is readily modifiable at the location of the PBX 10 and is alterable while the PBX is in an on-line operation.

The system administrator enters various commands at the system administration unit 24 to initiate modification of the call handling operations of telephonic calls terminating at PBX internal telephonic units 22. The system administrator creates a new or modified telescript at the system administration unit 24 for execution by the PBX central control processing unit 14 by selectively changing the procedural order of the call handling statements. Alternatively, the system administrator inserts additional particular call handling statements to the ordered set of statements to modify the telescript. The call handling feature script or telescript is altered by further selectively deleting certain call handling statements from the ordered set of statements of the defined telescript existing for a particular PBX internal telephonic unit 22. The modifications to the various telescripts are easily performed by the system administrator by simply altering the high level call handling statements at a command level as opposed to the necessity of reprogramming and recompiling the low level code at the instruction level as seen known PBX systems. The call handling feature scripts associated with the PBX internal telephonic units 22 are readily defined and altered by the system administrator or user of the system in a simple manner at the location of the PBX while the PBX is on-line to custom control the call handling features performed at each PBX telephonic unit.

Once the system administrator has changed the order or arrangement of the call handling statements by entering telescript commands at the system administration unit 24, the PBX central control processing unit 14 verifies that the changed order of call handling statements comply with the internal system parameters of the PBX 10. The central processing unit 14 creates a state machine derived from the changed ordered set of call handling statements if the central control processing unit verifies that the new or modified telescript compiles with the system parameter.

The system administrator stationed at the system administration unit 24 by altering the call handling statements of the telescript, in turn, selects the order of telephonic features for PBX calls directed to a PBX internal telephonic unit 22. The ordering the statements selectively relate to features which serve a plurality of PBX internal telephonic units 22 or groups of PBX units such as a key system group of units 25 or a hunt group of PBX units 23.

The ordered set of features for a defined telescript selectively modified by the system administrator include but are not limited to directing incoming calls to a voice mail group, directing calls to alternate PBX internal telephonic units if the dialed PBX unit is busy or programmed not to receive calls, directing a PBX unit to automatically call back a previous caller when the unit becomes idle, call forwarding of telephonic calls, and determining whether a PBX telephonic unit is busy or idle. The call handling features are identified through employment of the call handling feature statements and executed by the central control unit 14 stored in the main memory 16 of the PBX 10.

The selectively alterable telescripts associated with a PBX unit 22 also identify whether the PBX unit is a member of a preselected PBX hunt group 23 or a key system group 25. An intercept group is identifiable by the telescript for a particular PBX unit 22. Telephonic calls which cannot be completed or entirely processed are redirected to an intercept group or an alternative destination PBX unit for special processing. Control of the visual and audible indications to individuals stationed at PBX internal telephonic units 22 in response to received calls therein is defined in the telescript for the PBX units. An external remote host computer 44 alternatively is coupled to the central control processing unit 14 of the PBX 10. The host computer 44 selectively controls particular call handling functions for predetermined PBX internal telephonic units 22. A defined telescript directs the central control processing unit 14 to notify the host computer 44 of telephonic call flow at the predetermined PBX units 22 in order for the host computer to direct the logic at the central control unit and control the operations of the telephonic calls.

Other various features which are employed at the PBX 10 and are readily modified through the use of telescripts include but are not limited to: camp on busy, call waiting, day and night service, conference calling and distinctive ringing.

Individuals enter sign-in and sign-out codes at the respective PBX internal telephonic units 22. Preferably, the PBX internal telephonic units are standard Integrated System Digital Network (ISDN) basic rate interface (BRI) type interior telephonic units. The defined telescripts are associated with the individuals stationed at various PBX units 22. The telescripts correspond to a particular port coupled with a PBX internal telephonic unit 22 upon the sign-in of the individual at the particular PBX unit. Thus, the call handling feature scripts are preferably independent of the hardware associated with the individual PBX telephonic units 22. Staff individuals at a PBX interior telephonic units 22 place themselves in PBX termination groups upon sign-in at a PBX unit and remove themselves from the PBX termination group when signing-out at the PBX telephonic units.

The central control processing unit 14 of the PBX 10 determines if a particular call handling feature is available at the PBX internal telephonic unit 22 receiving a directed PBX call based on the read telescript defined for the individual stationed at the PBX unit. Generally while the private branch exchange can be implemented in numerous types and sizes of telecommunication systems, it is preferably implemented in a system of the types shown in U.S. Pat. No. 5,268,903 of Jones et al. entitled "Multichannel Telephonic Switching Network With Different Signaling Formats and Cross Connect/PBX Treatment Selectable For Each Channel", issued Dec. 7, 1993; U.S. Pat. No. 5,140,611 of Jones et al. entitled "Pulse Width Modulated Self-Clocking and Self-Synchronizing Data Transmission and Method for a Telephonic Communication Network Switching System", issued Aug. 18, 1992; U.S. Pat. No 5,127,004 of Lenihan et al. entitled "Tone and Announcement Message Code Generator for a Telephonic Switching System and Method", issued Jun. 30, 1992 and U.S. Pat. No 4,627,047 of Pitroda et al. entitled "Integrated Voice and Data Telecommunication Switching System", issued Dec. 2, 1986.

Referring now to FIG. 2, a detailed control representation for the user modifiable PBX terminating call feature controller 30 is shown having a main PBX telescript 28 and a plurality of call operation specific telescripts. The control of the main PBX telescript 28 and the associated call specific telescripts 58-72 operate through the PBX central control processing unit 14 with the associated control and handling software, telescripts and data stored in the main memory 16 of FIG. 1. Telephonic calls received from the external telephonic network 20 go through a routing telescript 31. The routing telescript 31 performs a translation on the destination digits dialed from the external telephonic unit 18 and receives a route instruction to terminate at a particular identified to PBX unit 22 destination from the routing processes in memory 16.

If the call is a internal call initiated by a PBX internal unit 22A the routing telescript 30 is not employed. The logical device handler for the individual calling at the internal PBX unit 22A, performs a translation on the dialed digits and receives a route to PBX termination destination instruction from the internal routing processes for PBX 10.

The main PBX termination telescript 28 is associated with every PBX termination unit 22 in the system. The main or primary PBX telescript 28 is implemented as a state machine which has a logical instance on every port of the multiport switch 12 of FIG. 1. For an incoming call to the switch 12 the primary PBX telescript 28, FIG. 2, starts executing at the beginning of the telescript. The variables controlled by the primary PBX telescript 28 include:

a. Is the PBX termination telephonic unit 22 busy or idle.

b. Is PBX termination unit 22 a member of a key system group 34.

c. Is the PBX unit 22 for termination a member of a hunt group 36.

d. Is the call forwarding call handling feature 38 activated.

e. Is the identified PBX termination unit 22 directing calls to the voice mail call handling feature 40.

f. Should the call be directed to the intercept feature 42.

g. Is the remote external host computer 44 to determine call disposition.

The main PBX telescript 28 is not limited to the above listed call handling controlling features and selectively includes alternative features provided by the system administrator. The main PBX telescript 28 selectively, uses all the call handling statements and variables used in other telescripts. If the PBX telephonic unit 22 for termination is a member of a key system group 34 the main PBX telescript 28 queues the key system group handler 46 to provide ringing and lamp indications to PBX units 22 which are members of the group that share appearances with the identified PBX termination telephonic unit 22.

If the PBX 10 is connected to an external host computer 44 and the primary PBX telescript 28 is programmed at the system administration unit 24, the primary PBX telescript 28 informs the host computer 44 of the call arrival and waits for directions for routing the call which is received from the host computer. If the host computer 44 does not respond the telescript continues after a control operator programmable delay time.

The PBX telescript control is independent of the physical PBX termination type unit 22 because a device driver 48 handles all unique hardware dependencies for various PBX telephonic units 22 and communicate with the main PBX telescript 28 by using generic events. The device map 50 under the control of the system administration unit 24 allows unique PBX telephonic units 22 to have translated generic telephony actions such as "on hook" "off hook" "flash" "dialed digits" etc.

The system administrator at administration unit 24 communicates with the central control processor 14, FIG. 1, and is able to program the main PBX telescript 28 in memory 16 using the internal-machine interface control programming 52, FIG. 2. All PBX Termination options are programmed by the system administrator through the human machine interface control 52 including the device specific information for PBX units at the device map 50 and configuration data 56 that specifies the operation of key system software including which internal telephonic PBX termination units 22 are in a key system group 34, and which directory numbers appear on each PBX termination unit and call appearance key. The options for primary and secondary appearances and the types and cadences of ringing applied at the identified PBX termination unit are provided.

PBX termination hunt group information 36 is also under the control of the system administrator at the administration unit 24. PBX hunt groups 36 are under the control of separate hunt group telescripts 58 and 60. The main PBX telescript 28 gives control to either the hunt group application telescript 58 or the PBX position hunt group telescript 60 which are provided programmed by the system administrator. The hunt group PBX telescripts 58 and 60 queue to the appropriate hunt group 36 which is controlled by the hunt group handler 62. The main PBX telescript 28 selectively transfers control to any system intercept telescript 64 as so provided by the system administrator 24.

The main PBX telescript 28 transfers control to the voice mail telescript 66 if enabled through the system administration unit 24. This enables the system administrator to determine at what point in a call that the calling party is given the option of leaving a voice mail message for a busy or not answering PBX termination telephonic unit 22.

The main PBX telescript 28 selectively transfers control to the extended ringing recall PBX telescript 68 if provisioned by the system administrator. The extended ringing recall telescript 68 allows for all the same options to be given to a caller that has been transferred to a PBX termination telephonic unit 22 that does not answer the call.

The main PBX telescript 28 further selectively transfers control to a call back telescript 70 if enabled through the system administration unit 24. The callback feature allows the calling party at an internal telephonic unit (i.e. PBX unit 22 or other type of internal telephonic unit at the multiport switch) to leave in the PBX 10, a request that when the previously busy called party is idle to be called back. The call back telescript allows the system administrator to direct the PBX central control processor 14 to provide the call handling desired when the call back feature attempts the call back but finds the party to be called back is busy again. The preferred action is to queue the caller again, to the call back queue, so that the identified terminating PBX unit 22 is automatically called back when it becomes idle.

The main PBX telescript selectively transfers control to the call park recall telescript 72. The called park recall telescript 72 is used when a call has been left in the call park state exceeding a preselected time period and is being directed to recall the party last talked to and it finds that party is presently busy.

The main PBX telescript 28 further selectively transfers control to the PBX operations telescript 74. The operations telescript 74 is used to provide the PBX operations that handle callers that have exhausted other system call handling operations. For example the operations telescript selectively plays general auto attendant messages, transfers a caller to a general voice mail box or sends a caller to intercept. The actions taken in the operations telescript are under the control of the system administrator at the system administration unit 24.

Referring now to FIG. 3, the key system operation is shown for the user modifiable PBX terminating call feature controller 30 of FIG. 2. The system administrator at system administration unit 24 through the human-machine interface control processing 52, programs the device dependent information into the device database 73 to control the specific PBX internal telephonic units 22. The system administrator further sets the configuration data 56 for the key system operation.

The primary or main PBX telescript 28 receives the telephonic call when the PBX 10 number decode has determined the particular call is identified to terminate at a PBX internal telephonic unit 22. The main PBX telescript 28 determines that the received call is to terminate to a particular PBX unit 22 that is member of a key system group 34 and it queues the caller to the key system group handler 46. The key system group handler 46 has the ringing control 55, lamp control 55 and termination status 57 for the associated PBX internal telephonic units 22 in order to control the visual and audible indications at the PBX units of a key system group upon receipt of a call at a unit.

Each of the device handlers 48 have queued themselves to their respective key system group 34 upon activation. Each device handler 48 further notifies the group handler 46 when it is available for ringing and when it is busy. The key system group handler 46 communicates with the main PBX telescript 28 in controlling the key system group 34.

The system administrator at administration unit 24 selectively controls the terminating and ringing of the key system group 34 through the main PBX telescript 28. The main telescript 28 determines that the PBX termination unit 22 is a member of a key system group 34 and if the PBX termination unit 22 is busy. If the position is idle and a member of a key system group 34 the telescript queues to the key system group handler 46 to ring the termination and any other terminations as defined by the key system group handler 46.

FIG. 4 is a bounce diagram of the events between various parts of the telecommunication system 11 for a call directed to terminate at a position for particular PBX telephonic unit 22. The left portion of FIG. 4 illustrates the events and state machines associated with the origination of the telephonic call from a calling telephonic unit 19. The calling telephonic unit is either an external telephonic unit 18, a PBX telephonic unit initiating a call for termination, or another internal telephonic unit such as an agent unit or supervisory unit (not shown) coupled with the multiport switch. The right portion of FIG. 4 shows the state machines and events

associated with the terminating side of the call at the PBX telephonic unit 22.

A new call is detected by the originating port logical device handler 76 and it sends a new call origination event to the call origination (COR) state machine 77. The call originating state machine 77 is in its idle state and goes to its dial state and prepares to receive the digits dialed at the calling telephonic unit 19. The transition to the new state and any functions needed to be performed in that transition are accomplished by processing modules executing in the PBX system memory 16, FIG. 1. A new call origination event is sent to the routing telescript state machine 31, FIG. 4, for it to analyze the received dialed digits once collected. Once the digits are received they are translated against digits stored in the system memory 16, FIG. 1, and the routing telescript state machine 31, FIG. 4, determines that the destination of the telephonic call is to an identified PBX termination telephonic unit 22. The routing telescript 31 sends a Route to PBX event to COR 77. COR 77 sends a new call origination event to the main PBX telescript state 78 machine of the calling party.

The PBX telescript state machine 78 sends a Request Information event requesting the Hunt Group Handler (HGH) 62 to determine if the called party at PBX termination unit 22 is an active hunt group member. The hunt group handler 62 responds with a not a member event. Since the called individual at PBX unit 22 is not in a hunt group the telescript 78 goes to its next statement.

Executing the next telescript statement the PBX telescript state machine 78 sends an Request Information event to the Key System Group Handler (KSH) 46 to determine the called PBX termination unit 22 is a member of a key system group. The KSH 46 responds with a Not A Member event. Since the called PBX termination unit 22 is not in a key system group the telescript goes to the next statement. From previous checks the PBX telescript state machine 78 knows that the line is a single telephonic line termination with no features active. The PBX telescript state machine 78 for the calling telephonic unit 19 sends the New PBX Call event to the logical device handler (LDH) 80 for the PBX unit 22 on the terminating side of the call to let it know that the new PBX call has arrived.

The called PBX unit LDH 80 sends a New PBX Call event to the Accept PBX Call (APC) state machine 81 which prepares the port to receive the call. The APC state machine 81 determines that the called party is idle and is available to be rung. The APC state machine 81 subsequently sends a port idle event to the PBX telescript 78 to let the PBX telescript know the APC can ring the PBX unit 22. The PBX telescript 78 sends a ring phone event to the APC 81 to initiate ringing of the PBX termination telephonic unit 22. APC 81 in turn sends a ring phone event to the hardware protocol handler 82 to actually ring the identified PBX unit 22. Upon the PBX termination unit 22 physically going off hook the telephone hardware protocol handler 82 detects the answer and sends a call answered event to the logical device handler 80 port of the PBX termination telephonic unit 22.

The called unit LDH 80 sends a call answered event to APC 81 to let it know the call has been answered. APC 81 then signals the calling party PBX telescript 78 that the call is answered. Calling party PBX telescript 78 returns to the idle state since its functions are complete. The call is completed by the called PBX unit APC 81 sending a connect call event to the called PBX unit call connected (CCD) state machine 83. The CCD state machine 83 sends the connect call event to the calling telephonic unit CCD 84 that the call is now connected. This causes the connection to be made and

the calling telephonic unit 19 and the called PBX termination telephonic unit 22 have a voice connection.

Referring now to FIGS. 5A–5D, in step 100, a call arrives at the switch 12, FIG. 1, of the PBX 10. In step 102, FIG. 5A, the central control processing unit 14 of the PBX 10 determines the routing of the received call by analyzing the digits dialed of the call. In step 104, the PBX 10 translates the received digits to determine if the call is a PBX telephonic call to terminate at an identified PBX position telephonic unit 22.

If the received call is not a PBX call, then in step 106, the call is handled by the existing non-PBX call processing software stored in the memory 16 of the system 10. If it is determined that the call is for a PBX termination position telephonic unit 22, then in step 108, the particular PBX call handling feature script or telescript in the PBX memory 16 associated with the PBX position telephonic unit 22 is installed on the port of the identified unit through the employment of capability mapping. For further details on the capability mapping feature and its associated operation, reference can be made to U.S. Pat. No. 5,365,581 of Baker et al. entitled "Telephonic Switching System With Automatic Port Assignment Capability and Method" issued Nov. 15, 1994.

In step 110, the first statement of the telescript installed on the port is read by the central control processing unit 14. In step 112, the PBX 10 determines if the first statement of the received telescript is one of the following statements: IF PBX, OFFER, SELECT, QUEUE UCD-HUNT GROUP, QUEUE BRIDGE APPEARANCE, QUEUE CALL BACK, ROUTE. The PBX 10 determines if the first statement is one of these listed statements by looking at the statement operand. If the first statement of the telescript for the terminating PBX unit 22 is not one of these new statements, then in step 114, the processing continues to execute the identified telescript statement.

If the first read statement is the IF PBX statement then in step 116, FIG. 5B, the procedure moves to step 148 to process the IF PBX statement. If the first statement of the telescript is the OFFER statement then in step 118, the PBX 10 accesses the circular hunt group data in memory 16. The OFFER statement indicates that the call is being offered to a circular hunt group. In step 118, the OFFER statement accesses the circular hunt group database in the main memory 16 to determine if there is an idle PBX termination unit 22 in the group. In step 120, the telescript sets the success or failure status for the OFFER statement in order that subsequent statements can determine if the call is able to terminate to a PBX unit member of a hunt group.

If the first statement of the telescript is the SELECT statement, then in step 122, the PBX 10 accesses staff entry and class of service information for the called party stationed at a PBX unit 22. The SELECT statement indicates that the PBX termination unit 22 is a single telephonic line unit and the telescript controls the ringing of the telephonic line. The staff entry for the identified individual at the PBX termination unit 22 is obtained in step 122, as well as class of service table memory being read from the staff entry of the individual signed-in at the PBX telephonic unit. The class of service provides an indication as to the type of ringing applied to the terminating PBX position telephonic unit 22. In step 124, the terminating PBX position telephonic unit 22 functions to ring as specified in its associated class of service table in the main memory 16.

If the first statement read is the QUEUE UCD-HUNT GROUP statement, then in step 126, FIG. 5B, then the PBX 10 obtains the hunt group number from the telescript. The

QUEUE UCD-HUNT GROUP statement indicates that the calling party is being terminated to a uniform call distribution group of PBX position telephonic units 22. In step 128, the central control processing unit 14 queues the call to identify the hunt group where the call waits in queue for an available PBX termination position telephonic unit 22.

If the first statement of the received telescript is the QUEUE BRIDGED APPEARANCE statement then the PBX 10 obtains ringing information from the key system database 34 in step 130, FIG. 5B. The QUEUE BRIDGED APPEARANCE statement indicates that the particular caller from an external telephonic unit 18 is terminating to a key set group of PBX position telephonic units 22. The key set configuration data is obtained from the key set database of the main memory 16. In step 132, the telescript processing commands the key set handler to ring the PBX position telephonic units 22 and illuminate call indication light emitting diode lamps on the unit console as specified in the key set database.

If the first received statement is the QUEUE CALL BACK statement, then in step 134, the caller from a calling telephonic unit 19, FIG. 4, is placed on QUEUE CALL BACK. The QUEUE CALL BACK statement indicates that the identified PBX position unit 22 is busy and ringing is not to be performed. The position PBX telephonic unit 22 has implemented a call back whereby the executing processes place the calling telephonic unit 19, FIG. 4, on a call back list to be subsequently called back when the PBX position telephonic unit 22 becomes idle.

If the first statement received is the ROUTE statement then in step 136, FIG. 5B, the call is to be processed by a different telescript and the processing goes to step 166, FIG. 5D, for the operation of the ROUTE statement.

In step 138, FIG. 5A, the central control processing unit 14 determines if the call has been connected to a termination through telescript processing. If the calling party has connected to a terminating device (i.e. voice mail, alternate party, call forwarded) then the telescript is closed in step 140. In step 142, the call processing control goes to the existing PBX call connected processing in the main memory 16. If the read statement does not result in a connection to a terminating port the flow goes to the next statement in step 144, the next statement is read and the control returns to step 112, FIG. 5A.

In step 148, of FIG. 5C, a determination is made that the next statement of the received telescript is the IF PBX statement. In step 150, the telescript processes running on the central control processing unit 14 of the PBX 10 determines the termination type being tested in this step for the particular call. In step 152, if the termination type is the originating party then the staff table database entry in memory 16 for the originator is accessed. In step 154, if the termination type is the individual PBX terminating unit 22, then the staff table database entry in memory 16 for the PBX terminating unit is accessed. In step 156, if the termination type is a system parameter, then the system parameter table database entry in memory 16 is accessed by the associated telescript. In step 158, a determination is made based on the information previously received from the database entry as to the parameter being checked in the IF PBX statement is either true or false. If the test is false the telescript software running on the PBX 10 prepares to process the next consecutive statement in the telescript at step 160. If the test is true, then in step 162, the telescript software running on the PBX 10 prepares to process a next statement specified in the telescript. Once the next statement is prepared to be processed then the flow returns to step 138 of FIG. 5A.

If the statement received is the ROUTE statement then in step 166, FIG. 5D, a determination is made as to which route command is received. If the received statement is a route extended ringing recall statement then in step 168, the system parameter database in memory 16 is accessed. The route extended ringing recall statement, indicates that the PBX telephonic unit 22 being rung has been ringing for a time period and the period has expired. The system parameter table in memory 16 is accessed to determine if the database has the feature activated. In step 170, a determination is made if the PBX 10 has the extended ringing recall feature. If the PBX 10 has the extended ringing recall feature, then in step 172, the recall telescript is installed on the port for the call terminating at the PBX unit 22 and control is transferred to the recall telescript. If the PBX 10 does not have the extended ringing recall feature active, then in step 174, the PBX ignores the time out and returns to step 138, of FIG. 5A. If the statement received is the route personal voice mail statement, then in step 176, FIG. 5D, the PBX 10 determines the directory number to send to the voice mail system 40, FIG. 2. The route personal voice mail statement indicates that the caller is to be routed to the voice mail telescript for leaving a voice mail message for the terminating party at the associated PBX position telephonic unit 22.

The telescript processes running at the central control processing unit 14 determines the mail box number to send to the voice mail system 40, FIG. 2, either from the telescript, the dial digits or the default general mail box number. In step 178, FIG. 5D, the telescript software running at the central control processing unit 14 installs the personal voice mail telescript and transfers control to the personal voice mail telescript.

If the statement received is the route call forward statement, then in step 180, the call forward loop counter is implemented. The receipt of a route call forward statement indicates that the terminating party at a PBX position telephonic unit 22 has preprogram call forwarding for the calls it received. The call forward loop counter is implemented in step 180. In step 182, the PBX 10 determines if the maximum number of call forwards have been exceeded. If the number of excessive call forwards exceeded the value specified in the telescript, then the call is sent to intercept processing. If the number of excessive call forwards have not exceeded the specified limit then the call continues.

In step 184, the call is transferred to the existing call intercept processing. In step 186, the new call forward information is stored in the system database in memory 16 for the associated call. In step 188, the processing returns to the start of the PBX telescript for a new directory number. A new instance of the PBX telescript is installed on the port and processing begins again at the first statement of the PBX telescript for the new destination of the call being forwarded.

Referring to FIG. 6, the user interaction and control flow for the creation and installation of a PBX telescript at a system administration unit 24 coupled with the central control processing unit 14 of the PBX 10 begins at step 200 in which the system administrator at the system administration unit 24 opens a new telescript.

The system administrator enters an "open telescript" Command on a keyboard (not shown) of the system administration unit 24 to create a new PBX telescript. In step 202, the system administrator enters the modified or new telescript statements of the telescript into the system administration unit 24 using the "enter telescript command". The system administrator is enabled to change the order and the particular type of telescript statements or ordered steps in

creating a new telescript while the PBX system is on-line, thereby selectively defining new call handling operation features for a particular PBX position telephonic unit 22. In step 204, FIG. 6, the PBX system 10 central control processing unit 14 determines and verifies if the statements entered conform with the syntax specified for the telescript action requested. If an error is detected then the procedural flow operating at the system administration unit 24 returns the system administrator to step 202 to enter a new telescript statement which meets the appropriate syntax.

If all the telescript action statements are entered which meet the correct syntax, then in step 206, the system administrator closes the telescript using the close telescript command. In step 208, the system administrator requests verification of the new telescript with the modified sequence of actions derived from the various statements of the telescript by entering a "verify telescript" command at the system administration unit 24. In step 210, the central control processing unit 14 coupled with the system administration unit 24 verifies that all the previously entered action statements operate together in accordance with the telescript parameters of the PBX 10. If the verification fails, then the system administrator or user is notified through the system administration unit 24 and the operation returns to step 202.

If the PBX 10 verifies the newly entered action statements resulting in a modified telescript, then in step 212, FIG. 6, the central control processing unit 14 creates the state machine for the new telescript from the partial state machines previously developed and coded for each action statement. This central control processing unit 14 creates an executable state machine though the processes derived of the various telescript action statements. In step 214, the system administrator enters the "activate telescript" command to install the state machine for use by the PBX 10. The central control processing unit 14 enters into the PBX system memory 16 data tables the location of the new PBX telescript. All newly received calls terminating to a PBX position telephonic unit 22 employ the new PBX telescript having the associated call handling features previously implemented on the system 10 in steps 200–214.

While a detailed description of the preferred embodiment of the invention has been given, it should be appreciated that many variations can be made thereto without departing from the scope of the invention as set forth in the appended claims.

I claim:

1. In a telecommunication system having a private branch exchange (PBX) with a multiport switch controlled by a central control processing unit and a PBX main memory for directing telephonic calls to terminate at identified ones of a plurality of PBX internal telephonic units placed at predetermined positions and connected with the multiport switch, the improvement comprising:

means for storing a call handling feature script defining call handling operations for telephonic calls directed to an identified PBX internal telephonic unit and in which the call handling feature script includes an ordered set of call handling statements with each of the statements defining a particular call servicing operation; and

means for modifying the call handling feature script in response to the entry of user commands via a data entry terminal coupled with the central control processing unit of the PBX to alter the call handling operations of telephonic calls terminating at the identified PBX internal telephonic unit while the PBX is actively on-line and without requiring the reprogramming and recompiling of programmed code.

2. The telecommunication system of claim 1 in which the modifying means is capable of at least one of adding particular call handling statements to the ordered set of the call handling feature script and deleting particular call handling statements from the ordered set of the call handling feature script.

3. The telecommunication system of claim 1 in which the modifying means includes means for changing the order of the call handling statements of the call handling feature script.

4. The telecommunication system of claim 3 including means for verifying that the changed order of the call handling statements comply with the internal system parameters of the PBX.

5. The telecommunication system of claim 4 including means for creating a state machine derived from the changed ordered set of call handling statements in response to said verifying means.

6. The telecommunication system of claim 1 including a primary PBX telescript having a predefined set of call handling statements associated with each of the plurality of PBX internal telephonic units and in which the primary PBX telescript is implemented as a state machine having a logical instance on the ports of the multiport switch for controlling the call handling operations at the PBX internal telephonic units.

7. The telecommunication system of claim 6 in which said primary telescript determines whether the identified PBX internal telephonic unit is at least one of

(a) a busy unit,

(b) a member of a preselected key system group.

(c) a member of a preselected hunt group,

(d) a unit directing telephonic calls to voice mail,

(e) a unit call forwarding telephonic calls,

(f) a unit which intercepts telephonic calls which cannot be processed and has said calls redirected to an alternative destination,

(g) a unit having a host computer coupled with the PBX for determining call disposition, and

(h) an idle unit.

8. The telecommunication system of claim 7 in which the primary telescript determines a plurality of (a)–(h).

9. The telecommunication system of claim 1 in which the modifying means is capable of performing modifications to the call handling feature script while the PBX is on-line to actively direct telephonic calls to terminate at the identified ones of the PBX internal telephonic units.

10. The telecommunication system of claim 9 in which said data entry terminal is a system administration unit electrically coupled with the central control processing unit for entering said commands to initiate the alteration of said call handling operations for the PBX internal telephonic unit at the PBX while the PBX is on-line to continue the PBX to actively direct telephonic calls to terminate at the identified ones of the PBX internal telephonic units.

11. The telecommunication system of claim 1 including means for selectively controlling the call handling operations to perform at least one of

(a) directing telephonic calls incoming to the PBX telephonic unit to a voice mail group,

(b) directing telephonic calls incoming to the PBX telephonic unit to an intercept group,

(c) directing telephonic calls incoming to the PBX telephonic unit to a PBX hunt group,

(d) directing telephonic calls incoming to the PBX telephonic unit to alternate PBX internal telephonic units,

(e) directing the PBX telephonic unit to automatically call back a plurality of callers,

(f) providing visual and audible indications at PBX internal telephonic units of a group in response to receipt of telephonic calls at the PBX internal telephonic units, and

(g) notifying a host computer coupled with the PBX of calls directed to the PBX internal telephonic unit.

12. The telecommunication system of claim 11 including means for selectively controlling a plurality of the call handling operations of (a)–(g).

13. The telecommunication system of claim 12 including means for determining if one of the plurality of call handling operations is available at the PBX internal telephonic unit.

14. In a telecommunication system having a private branch exchange (PBX) with a multiport switch controlled by a central control processing unit and a PBX main memory for directing telephonic calls to terminate at identified ones of a plurality of PBX internal telephonic units placed at predetermined positions and connected with the multiport switch, the improvement being a method of modifying PBX terminating call features, comprising the steps of:

storing a call handling feature script defining call handling operations for telephonic calls terminating at an identified PBX internal telephonic unit and in which the call handling feature script includes an ordered set of call handling statements with each of the statements defining a particular call servicing operation; and

modifying the call handling feature script in response to the entry of user commands at via a system administration unit coupled with the central control processing unit to alter the call handling operations of telephonic calls terminating at the identified PBX internal telephonic unit and in which such user commands are entered while the PBX is actively on-line to continually direct telephonic calls to terminate at the identified ones of the PBX internal telephonic units without requiring the reprogramming and recompiling of programmed code.

15. The method of claim 14 including the step of changing the order of the call handling statements of the call handling feature script.

16. The method of claim 15 including the step of verifying that the changed order of the call handling statements comply with the internal system parameters of the PBX.

17. The method of claim 16 including the step of creating a state machine derived form the changed ordered set of call handling statements in response to the verification of the changed ordered set of call handling statements comply with the internal system parameters of the PBX.

18. The method of claim 14 including the step of selectively controlling the call handling operations to perform at least one of

(a) directing the telephonic calls incoming to the PBX telephonic unit to a voice mail group,

(b) directing telephonic calls incoming to the PBX telephonic unit to an intercept group,

(c) directing telephonic calls incoming to the PBX telephonic unit to a PBX hunt group,

(d) directing telephonic calls incoming to the PBX telephonic unit to alternate PBX internal telephonic units,

(e) directing the PBX telephonic unit to automatically call back a plurality of callers,

(f) providing visual and audible indications at PBX internal telephonic units of a group in response to receipt of telephonic calls at the PBX internal telephonic units, and

(g) notifying a host computer coupled with the PBX of calls directed to the PBX internal telephonic unit.

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United States Patent [19]

Bartholomew et al.

[11] **Patent Number: 6,167,119**

[45] **Date of Patent: Dec. 26, 2000**

[54] **PROVIDING ENHANCED SERVICES THROUGH SIV AND PERSONAL DIAL TONE**

[75] Inventors: **Dale L. Bartholomew**, Vienna; **Robert D. Farris**, Sterling, both of Va.; **Alexander I. McAllister**, Silver Spring; **Michael J. Strauss**, Potomac, both of Md.

[73] Assignee: **Bell Atlantic Network Services, Inc.**, Arlington, Va.

[21] Appl. No.: **09/006,033**

[22] Filed: **Jan. 12, 1998**

Related U.S. Application Data

[63] Continuation-in-part of application No. 08/828,959, Mar. 8, 1997, Pat. No. 5,978,450, and a continuation-in-part of application No. 08/904,936, Aug. 1, 1997, Pat. No. 6,038,305, and a continuation-in-part of application No. 08/997,505, Dec. 23, 1997, Pat. No. 6,101,242.

[51] **Int. Cl.**[7] **H04M 3/42**

[52] **U.S. Cl.** **379/88.04**; 379/207

[58] **Field of Search** 379/67.1, 88.01, 379/88.02, 88.03, 88.04, 88.22, 88.26, 201, 207

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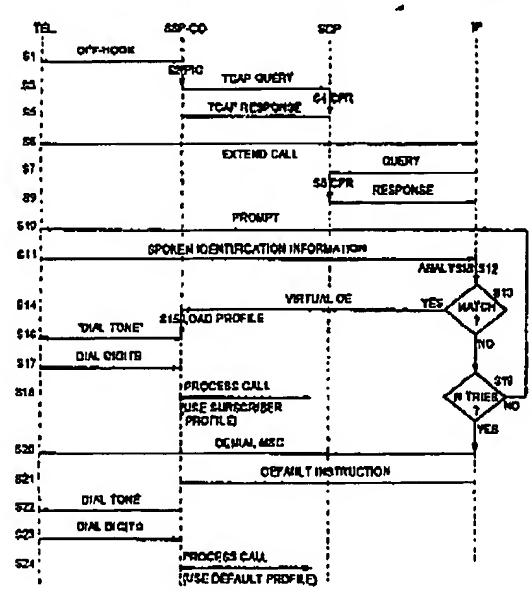
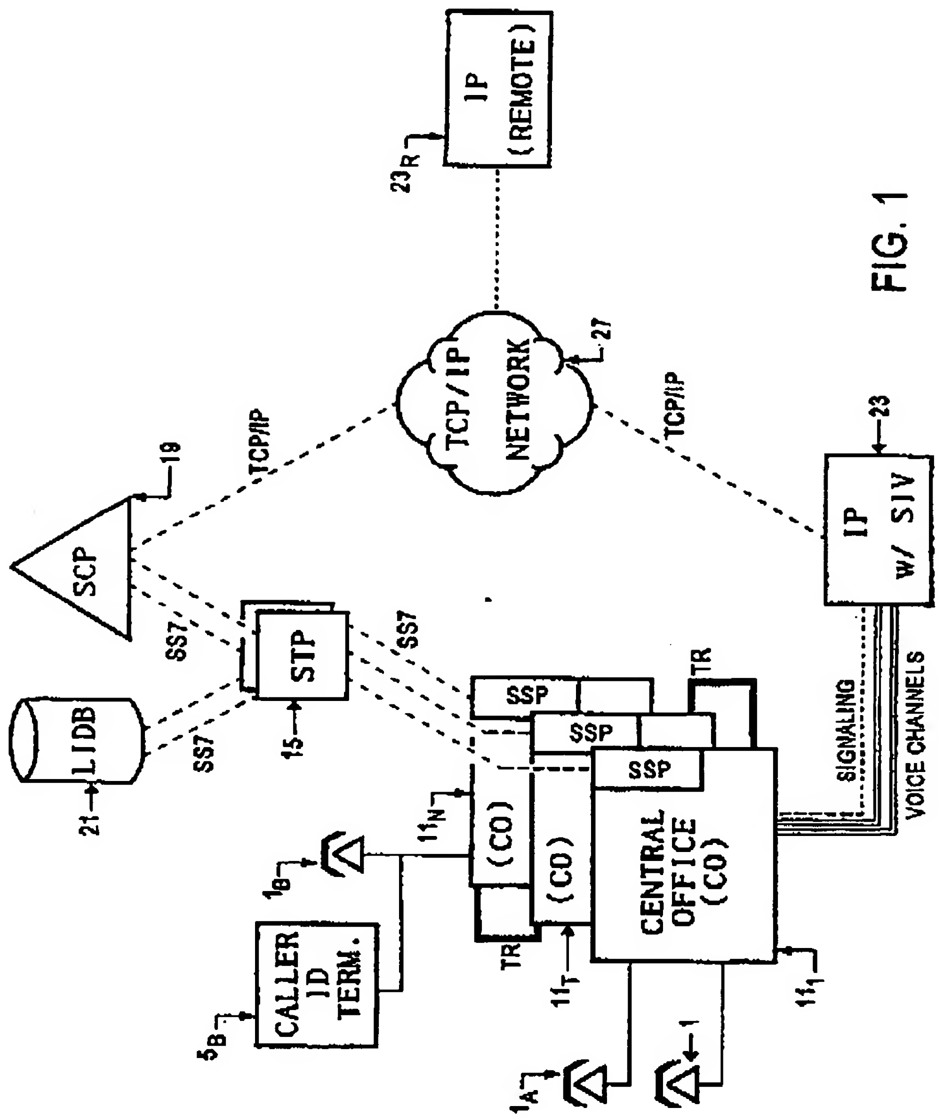
Primary Examiner—Scott L. Weaver
Attorney, Agent, or Firm—McDermott, Will & Emery

[57] **ABSTRACT**

An intelligent telephone network provides personalized communication services based on subscriber prescribed double speech signal processing of utterances of both calling and answering parties on a subscriber line having multiple subscribers with a single directory number. Specifically, when one of the multiple subscribers has personalized voice mail service and a busy/no answer call is received, the network uses speech processing of an utterance of the calling party to identify a customer service profile of the called party in the terminating switch. This contains instructions inviting storage of a voice message left by the caller. Upon the subscriber going off-hook, the customer profile of the subscriber line is installed in the switch. The subscriber transmits an utterance and the personalized customer profile of the subscriber is identified by a virtual office equipment number and the profile installed in the switch. The subscriber may then retrieve the stored message.

41 Claims, 7 Drawing Sheets

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SCP
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IP w/ SIV
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15
LIDB
21
SSP
SSP
SSP
TR
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SIGNALING
VOICE CHANNELS
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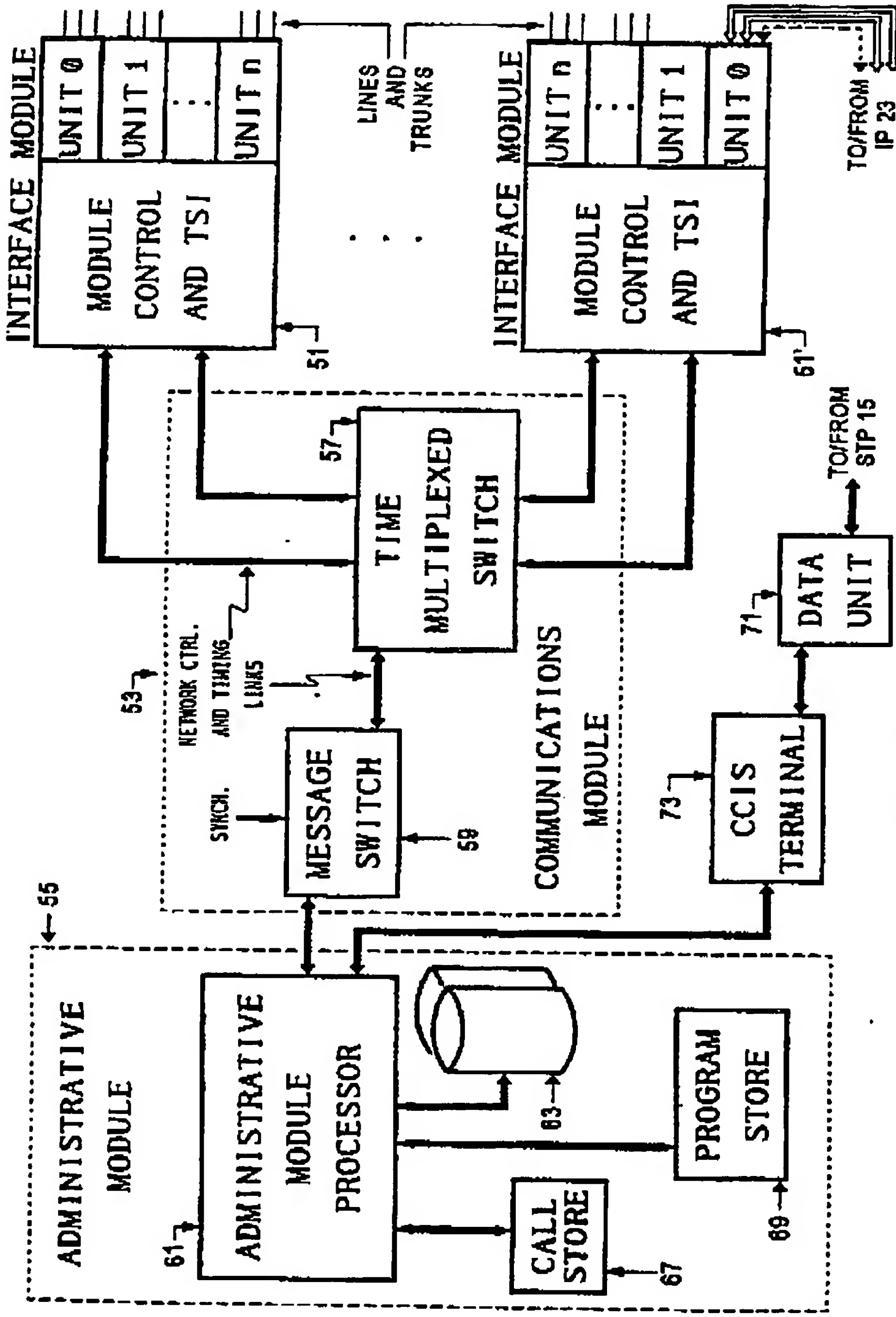
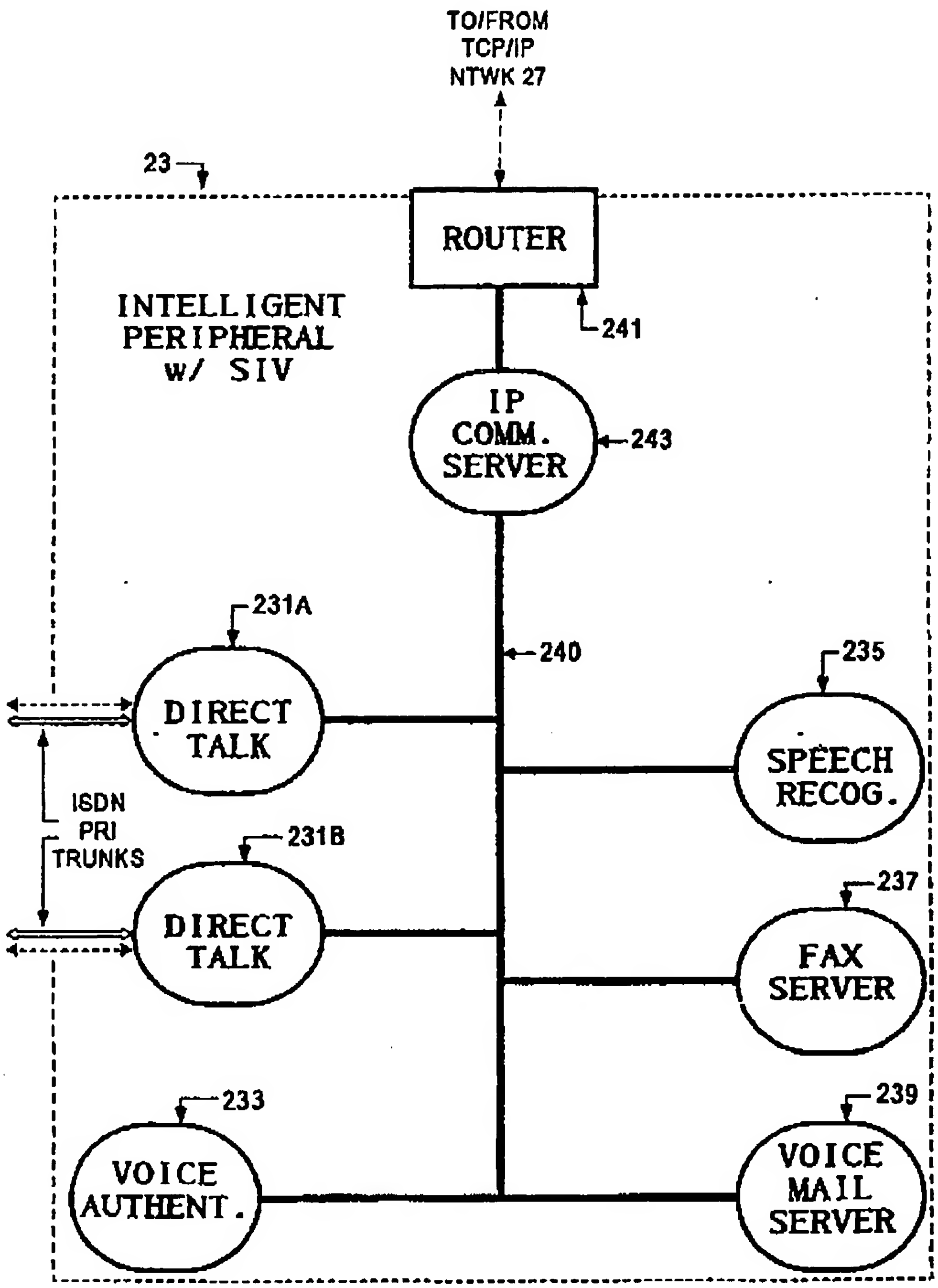
CALLER ID TERM.
5B

FIG. 1



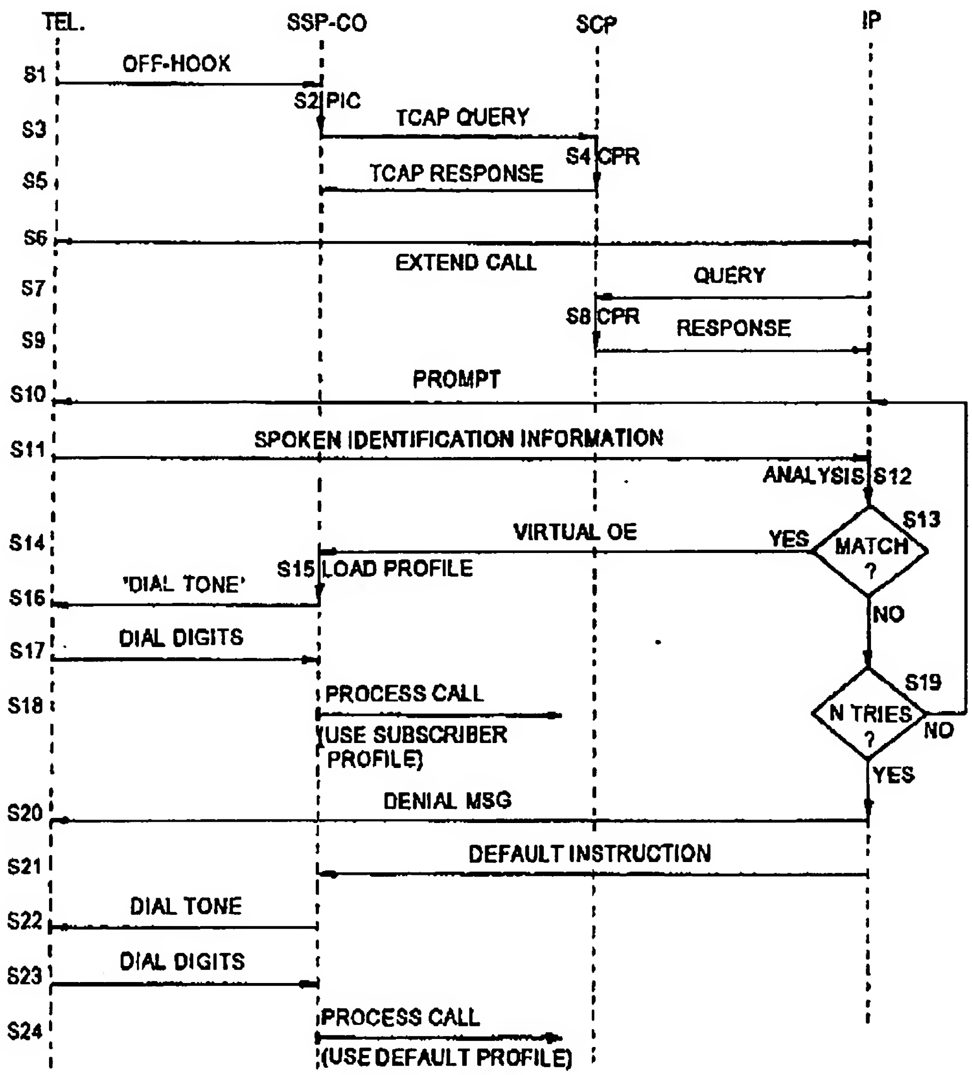
INTERFACE MODULE
UNIT 0
UNIT 1
UNIT n
MODULE CONTROL AND TSI
51
LINES AND TRUNKS
INTERFACE MODULE
UNIT n
UNIT 1
UNIT 0
MODULE CONTROL AND TSI
TO/FROM IP 23
51'
57
TIME MULTIPLEXED SWITCH
TO/FROM STP 15
53
NETWORK CTRL. AND TIMING LINKS
71
DATA UNIT
SYNCH.
MESSAGE SWITCH
59
COMMUNICATIONS MODULE
73
CCIS TERMINAL
55
ADMINISTRATIVE MODULE
ADMINISTRATIVE MODULE PROCESSOR
61
63
PROGRAM STORE
69
CALL STORE
67

FIG. 2



TO/FROM TCP/IP NTWK 27
23
ROUTER
241
INTELLIGENT PERIPHERAL w/ SIV
IP COMM. SERVER
243
231A
240
235
DIRECT TALK
SPEECH RECOG.
ISDN PRI TRUNKS
231B
237
DIRECT TALK
FAX SERVER
233
239
VOICE AUTHENT.
VOICE MAIL SERVER

FIG. 3



TEL.
SSP-CO
SCP
IP
S1
OFF-HOOK
S2
PIC
S3
TCAP QUERY
S4
CPR
S5
TCAP RESPONSE
S6
EXTEND CALL
S7
QUERY
S8
CPR
S9
RESPONSE
S10
PROMPT
S11
SPOKEN IDENTIFICATION INFORMATION
ANALYSIS
S12
S13
MATCH ?
YES
NO
S14
VIRTUAL OE
S15
LOAD PROFILE
S16
'DIAL TONE'
S17
DIAL DIGITS
S18
PROCESS CALL
(USE SUBSCRIBER PROFILE)
S19
N TRIES ?
NO
YES
S20
DENIAL MSG
S21
DEFAULT INSTRUCTION
S22
DIAL TONE
S23
DIAL DIGITS
S24
PROCESS CALL
(USE DEFAULT PROFILE)

FIG. 4A

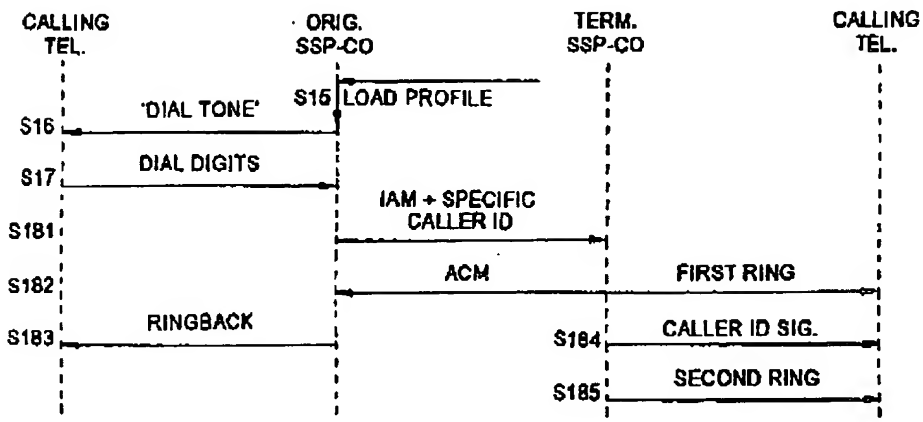

CALLING TEL.
ORIG. SSP-CO
TERM. SSP-CO
CALLING TEL.
S15
LOAD PROFILE
S16
'DIAL TONE'
S17
DIAL DIGITS
S181
IAM + SPECIFIC CALLER ID
S182
ACM
FIRST RING
S183
RINGBACK
S184
CALLER ID SIG.
S185
SECOND RING

FIG. 4B

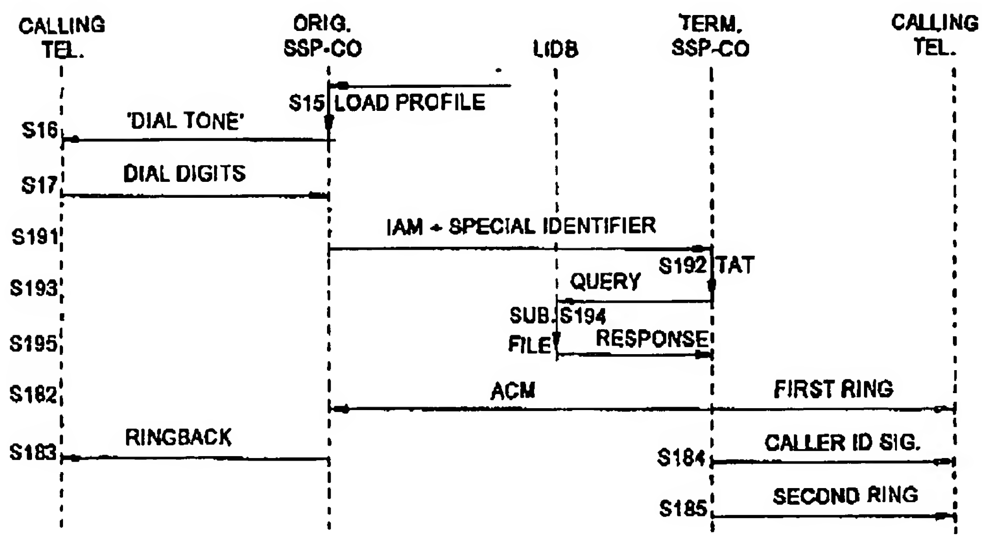

CALLING TEL.
ORIG. SSP-CO
LIDB
TERM. SSP-CO
CALLING TEL.
S15
LOAD PROFILE
S16
'DIAL TONE'
S17
DIAL DIGITS
S191
IAM + SPECIAL IDENTIFIER
S192
TAT
S193
QUERY
SUB. FILE
S194
S195
RESPONSE
S182
ACM
FIRST RING
S183
RINGBACK
S184
CALLER ID SIG.
S185
SECOND RING

FIG. 4C

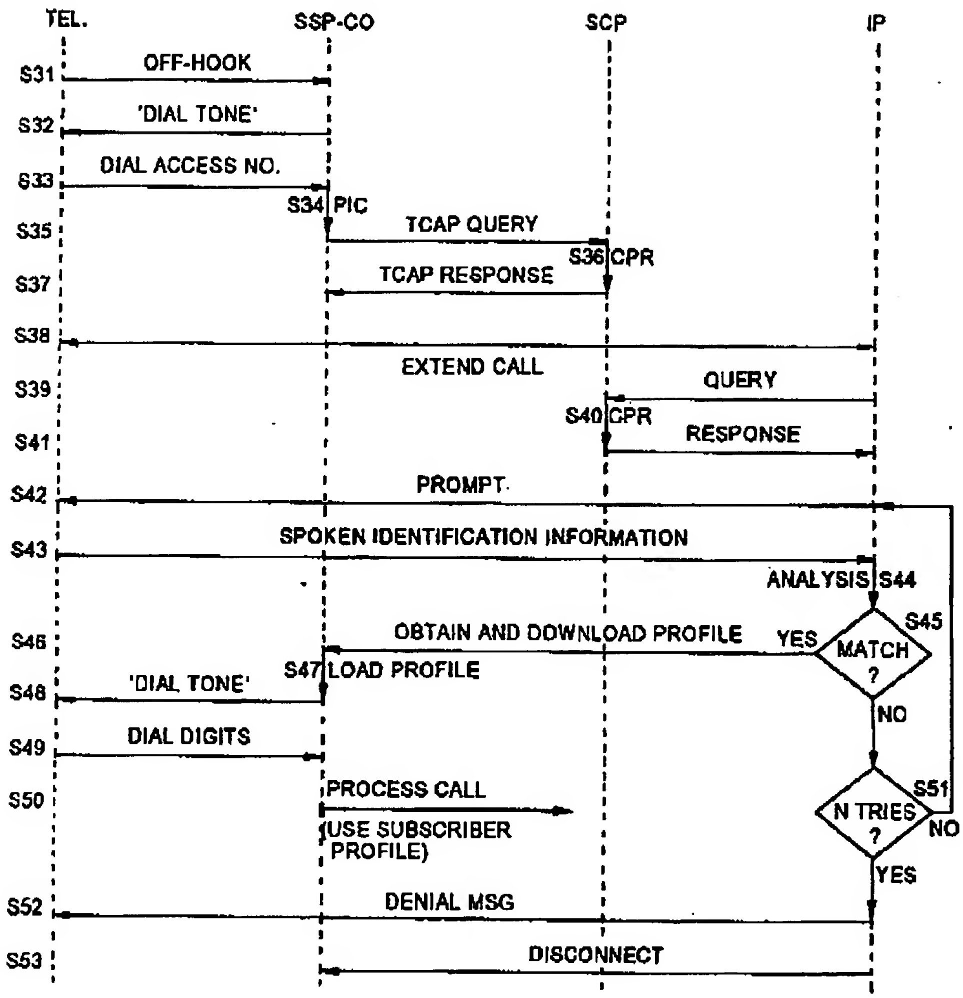

TEL.
SSP-CO
SCP
IP
S31 OFF-HOOK
S32 'DIAL TONE'
S33 DIAL ACCESS NO.
S34 PIC
S35 TCAP QUERY
S36 CPR
S37 TCAP RESPONSE
S38 EXTEND CALL
S39 QUERY
S40 CPR
S41 RESPONSE
S42 PROMPT
S43 SPOKEN IDENTIFICATION INFORMATION
ANALYSIS S44
S45 MATCH ?
YES
NO
S46 OBTAIN AND DOWNLOAD PROFILE
S47 LOAD PROFILE
S48 'DIAL TONE'
S49 DIAL DIGITS
S50 PROCESS CALL
(USE SUBSCRIBER PROFILE)
S51 N TRIES ?
NO
YES
S52 DENIAL MSG
S53 DISCONNECT

FIG. 5

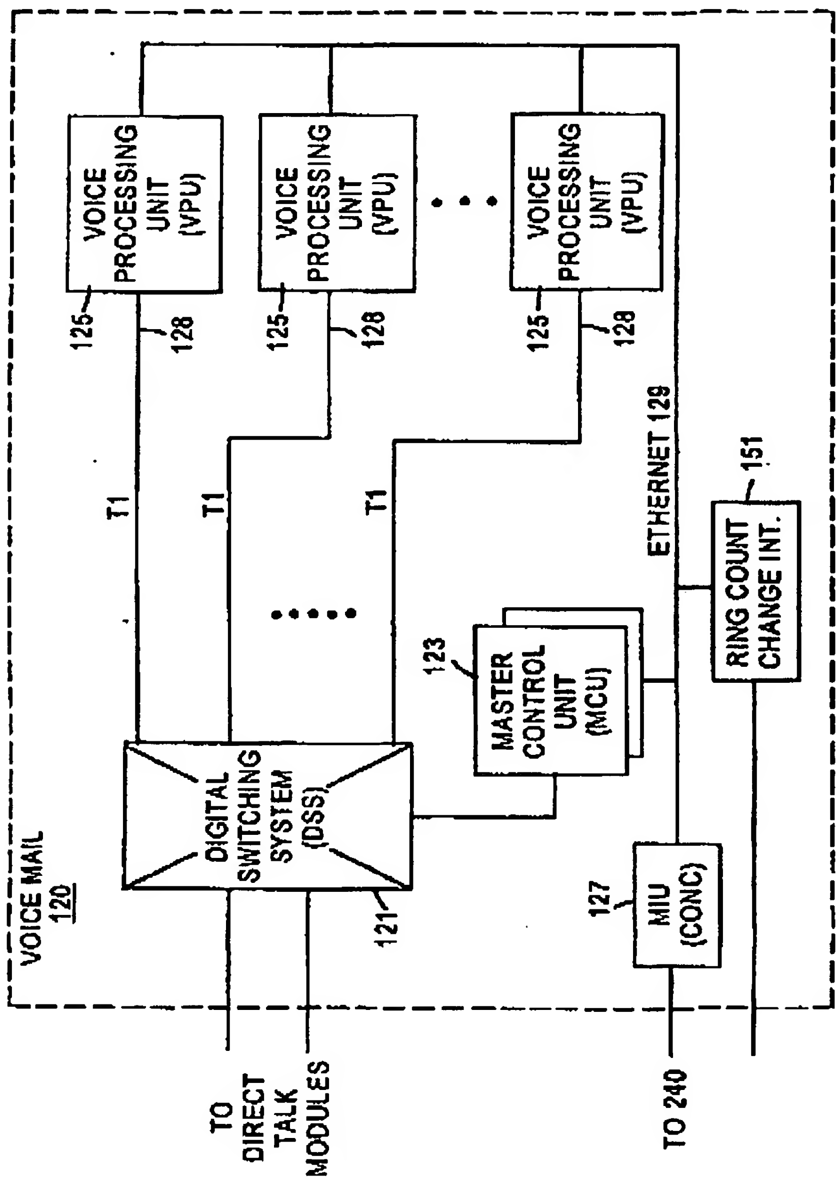

VOICE MAIL 120
VOICE PROCESSING UNIT (VPU)
VOICE PROCESSING UNIT (VPU)
VOICE PROCESSING UNIT (VPU)
125
128
125
128
125
128
T1
T1
T1
ETHERNET 129
151
RING COUNT CHANGE INT.
123
MASTER CONTROL UNIT (MCU)
DIGITAL SWITCHING SYSTEM (DSS)
121
127
MIU (CONC)
TO DIRECT TALK MODULES
TO 240

FIG. 6

PROVIDING ENHANCED SERVICES THROUGH SIV AND PERSONAL DIAL TONE

CROSS-REFERENCE TO RELATED APPLICATIONS

This is a continuation-in-part of U.S. patent applications Ser. No. 08/828,959, filed Mar. 8, 1997, now U.S. Pat. No. 5,978,450; 08/904,936, filed Aug. 1, 1997, now U.S. Pat. No. 6,038,305; and 08/997,505, filed Dec. 23, 1997, now U.S. Pat. No. 6,101,242 the disclosures of which are incorporated entirely by reference.

TECHNICAL FIELD

The present invention relates to personalized telecommunications service, preferably offered through an intelligent telephone network. In particular, the present invention relates to the identification of one or both calling and answering speakers to control processing of the communication. Enhanced services are provided on a personalized basis to multiple subscribers using the same line to terminal equipment.

Acronyms

The written description uses a large number of acronyms to refer to various services, messages and system components. Although generally known, use of several of these acronyms is not strictly standardized in the art. For purposes of this discussion, acronyms therefore will be defined as follows:

Address Complete Message (ACM)
Advanced Intelligent Network (AIN)
ANswer Message (ANM)
Automatic Number Identification (ANI)
Call Processing Record (CPR)
Central Office (CO)
Common Channel Interoffice Signalling (CCIS)
Data and Reporting System (DRS)
Destination Point Code (DPC)
Generic Data Interface (GDI)
Initial Address Message (IAM)
Integrated Service Control Point (ISCP)
Integrated Services Digital Network (ISDN)
ISDN User Part (ISDN-UP)
Intelligent Peripheral (IP)
Line Identification Data Base (LIDB)
Multi-Services Application Platform (MSAP)
Office Equipment (OE)
Origination Point Code (OPC)
Personal Communications Service (PCS)
Plain Old Telephone Service (POTS)
Point in Call (PIC)
Personal Identification Number (PIN)
Primary Rate Interface (PRI)
Public Switched Telephone Network (PSTN)
Service Control Point (SCP)
Service Creation Environment (SCE)
Service Management System (SMS)
Service Switching Point (SSP)
Signaling System 7 (SS7)
Signaling Point (SP)
Signaling Transfer Point (STP)
Simplified Message Desk Interface (SMDI)
Speaker Identification/Verification (SIV)
Terminating Attempt Trigger (TAT)
Time Slot Interchange (TSI)
Traffic Service Position System (TSPS)

Transaction Capabilities Applications Part (TCAP)
Transmission Control Protocol/Internet Protocol (TCP/IP)

BACKGROUND ART

Today the public switched telephone network (PSTN) and other telephone networks such as cellular systems provide most telephone services based on number identification of the telephone set or line that each party uses. Services are personalized only to the extent that a party uses the same line and/or instrument. For example, a person typically has one set of service features and billing options available via a telephone on the person's desk at the office, another set of service features and billing options available via the telephone line to their home and perhaps a third set of service features and billing options available via a wireless telephone (e.g. cellular or personal communications service (PCS)). The networks process calls to and from each of these different subscriber telephones based on a separate telephone number. Also, a caller may use personalized billing options by using a calling card, but often the input operations for calling card service are overly complex. With the exception of calling card billing, each person using a particular telephone typically can only access those service features and billing options associated with the particular line or telephone instrument.

The proliferation of services causes subscribers inconvenience. For example, circumstances arise in which a subscriber may want a feature or billing option normally associated with one line or instrument, such as the office telephone, when they are in fact using a different line or instrument such as their home or PCS telephone. Alternatively, two or more persons using one telephone or line often want different sets of service options. Also, the extreme increase in demand for telephone services is rapidly exhausting the capacity of the network, particularly in terms of the telephone numbers available under the current numbering plan.

A number of specific solutions have been proposed for individual problems, such as work at home and/or transfer of service to new locations) as an individual travels. However, each of these solutions is limited or creates its own new problems.

For example, U.S. Pat. No. 4,313,035 to Jordan et al. discloses a method of using an intelligent network to provide a 'follow-me' type service through multiple exchanges of the switched telephone network using an AIN type of telephone system architecture. Each subscriber to the locator service has a unique person locator telephone number. To access the system to update data in a service control database, the subscriber dials 0700 and his unique person locator telephone number. The telephone switching office routes the call to a traffic service position system (TSPS) which prompts the caller (e.g. provides an additional dial tone) and receives further digits from the subscriber. The subscriber inputs a three digit access code, indicating the type of update call, and a four digit personal identification number. If calling from the remote station to which the subscriber wishes his calls routed, the local switching office forwards the line identification number of that station to the TSPS. The TSPS forwards the dialed information and the line identification to the data base for updating the particular subscriber's location record. A caller wishing to reach the subscriber dials the subscriber's unique person locator number. A telephone switching office sends the dialed number to the central database. The database retrieves the stored completion number for the called subscriber and forwards that number back to the switching office to complete the call.

The Jordan et al. approach allows calls to follow the subscriber to each new location, but the subscriber must have a unique telephone number for this service. Each station that receives a call also must have a unique telephone number. As such, the Jordan et al. approach actually exacerbates the shortage of telephone numbers. Also, Jordan et al. rely on subscriber input of identification numbers. Subscribers often find this inconvenient, and this technique is often prone to number entry errors.

U.S. Pat. No. 4,899,373 to Lee et al. discloses a system for providing special telephone services to a customer on a personal basis, when the customer is away from his or her home base or office. The personalized services are provided in a multiple exchange office environment, using a central database for feature control. The nationally accessible central database system stores feature data in association with personal identification numbers. A subscriber wishing to use his personalized features while away from home base dials a special code and presents the personal identification number. The exchange transmits a query to the central database, and the corresponding feature data is retrieved from the database. The database forwards the feature data to the exchange, and the exchange stores the received feature data in association with the station from which the request was initiated. Subsequently, the exchange accesses the downloaded feature data to provide telephone service corresponding to the subscriber's personalized telephone features via the station the subscriber is currently operating from. A temporary office arrangement may be established in which the personalized features will be immediately available on incoming and outgoing calls for a period of time specified by the subscriber.

U.S. Pat. No. 5,206,899 to Gupta et al. pertains to a system wherein a subscriber can assign desired characteristics to any "target station" which is an active telephone accessible to a telecommunications network. A call thereafter that originates from the target station can use customized features, such as account code dialing and corporate billing arrangements. Initially, a service profile is created and stored for each subscriber and contains information describing desired features and billing options. The characteristics of a particular target station are changed by an activation process that can be initiated from any location. Automatic number identification (ANI) information associated with the target station is entered into an ANI trigger table in an intelligent switch, and the service profile is loaded into a database. When a call originates from the target station, information in the database is applied to the switch to provide the desired characteristics. An example of one of the features is when an employee of company X wishes to make business related calls from his/her telephone, the call has the characteristics of a call made from the office by a special billing arrangement.

Like Jordan, the Lee et al. and Gupta et al. systems depend on a dialed number entry by the subscriber to activate the service. Also, the Lee et al. and Gupta et al. systems do not provide a simple manner for more than one subscriber to obtain personalized service over the same telephone line. In Lee et al., during the period when the switch stores the roaming subscriber's profile in association with the line, all calls are processed based on that one profile. Similarly, in Gupta et al., while the ANI trigger is set against the line, all outgoing calls cause database access and use of the subscriber's profile in the database. There is no way to fall back on the normal profile for that line unless and until the service for the roaming subscriber is cancelled with respect to that one line.

U.S. Pat. No. 5,247,571 to Kay et al. discloses an Area Wide Centrex service provided by an advanced intelligent telephone network. The service provides centrex features, such as extension dialing, to multiple locations. The Kay et al. Patent also suggests a Work-at-Home feature. This feature allows the home telephone line to selectively operate as a residential line or as a Centrex business line, on a call-by-call basis. For a business call, the user would preface each call with an access indicator to identify a business call. When an outgoing call from the home line lacks the access indicator, the network processes the call as a standard residential call.

The Work-at-Home feature in the Kay et al. system requires only dialing of a code before each outgoing business call. However, the Kay et al. approach requires that the business profile is stored in association with the home line before the subscriber makes the call. The subscriber can use the Centrex billing and service features from the business account only from a home telephone previously associated with the business line. The subscriber can not use the billing and service features from the business account from any randomly selected telephone. Also, from the home line, a person can either use the normal residential profile service or the pre-defined business profile service. There is insufficient flexibility to enable a wider range of services for multiple subscribers through the one line.

U.S. Pat. No. 5,422,936 to Douglas J. Atwell, issued Jun. 6, 1995, describes an Enhanced Message Service Indication. For a number of years, telephone companies have been providing a service which assigned two or more directory numbers per line and corresponding distinctive ringing signals. One of the telephone switch vendors refers to this feature as "Multiple Directory Numbers per Line" or "MDNL." This patent provides a system for providing voice mail service in a MDNL situation. The system is effective in serving its intended purpose but assumes the assignment of one directory or telephone number for each subscriber or service. As previously stated the current demand for telephone services is rapidly exhausting the capacity of the network, particularly in view of the telephone numbers available under the current numbering plan.

An increasingly popular telephone services is caller identification or 'caller ID'. The telephone network identifies the telephone number associated with the line or instrument used by the calling party and supplies the number and or the name to a display device at the called customer's premises.

Subscribers having ISDN service receive caller ID data, for display at the time of an incoming call, in the form of a data message which the end office switch transmits over the D-channel. For analog telephone customers, however, existing caller ID utilizes in-band transmission technology similar to that described in U.S. Pat. Nos. 4,582,956 and 4,551,581 to Doughty. In such a system, the end office switch connected to the called party's line transmits directory number data for the calling party's telephone line as frequency shift keyed (FSK) data inserted in the silent interval between ringing signal pulses applied to the called party's line. The receiving apparatus includes a line interface unit, a converter, a control circuit and a display unit. A frequency shift keyed (FSK) signal representing the special service information is filtered from the ringing signals by the line interface unit. The converter detects the FSK signal and demodulates the special service information from the FSK signal. Following detection of the FSK signal, the control circuit receives and stores the special service information. The stored information is periodically sent to the display unit to begin exhibiting thereof during the silent interval before the next ringing signal.

The local telephone exchange carriers have recently begun offering an enhanced form of caller ID, sometimes referred to as 'Caller ID Deluxe' service. This enhanced service utilizes AIN type call processing to access a Line Information Database (LIDB) to translate the calling party's directory number into name data. The end office switch forwards the name data and the normal caller ID telephone number as FSK encoded data inserted in the silent intervals between ringing signals.

The LIDB database includes a single listing for each telephone line and translates each number into a single name, typically the name of the party identified as the customer or subscriber for billing purposes. In fact, the LIDB database provides this single translation even for calls from one line having multiple telephone numbers. Consider an example in which a family has one line with two numbers and a distinctive ringing service. The first number is used for the family as a whole, and the second number is used for a teenage son or daughter. The distinctive ringing allows people in the household to know whether or not each call is for the teenager. On outgoing calls, however, the end office switch always identifies the line by the primary number (the family's number), and the LIDB database always provides the name of the billing subscriber, e.g. the father's name. As a result, when the teenager calls a friend, the friend will receive the main number and possibly the father's name. If the friend calls back using the information from his caller ID display terminal, the friend calls the family's main number, not the teenager's number.

Also, the above discussed examples of prior suggestions to customize services have not adapted the caller identification to correspond to the actual party using the telephone on the outgoing call. For example, in a system like that of Lee, Gupta or Kay, the caller might use features and billing options associated with her personalized or work service, but any such calls would produce a caller ID display identifying the number of the station from which she originated the call. If the called party subscribed to the name type enhanced caller ID, the network would provide a name associated with that telephone number, not the name of the actual calling party.

U.S. Pat. Nos. 4,961,217 and 4,759,056 disclose a card based system for providing personalized features, including caller name display. Each user has a "portable memory device" in the form of an identification card bearing personal information including identification information. When initiating a call, the user inserts the card in the calling station, and information from the card is transmitted to the central switching system. In one embodiment, the switching system translates the identification information from the card to produce a textual representation of the calling party's name and transmits that information to a called terminal for display. Although this system does provide a name display identifying the actual called party, the system requires the use of the identification card and specialized calling terminals for reading the information from the cards.

Another enhanced service which has become extremely popular is so called Voice Mail service. Voice mail is a service which may be considered a custom calling service and normally includes in its operation the use of call forwarding. Voice mail has become commonplace not only in business usage but also on an individual telephone service subscriber basis through Centrex service from a central office. A voice mail system is a specialized computer that stores messages in digital form on a fixed disk. The voice is generally digitized, usually at a much slower rate than the 64 Kb/s signal the central office uses in its switching network. The digitized voice is compressed and stored on a hard disk that maintains the voice mail operating system, system prompts, and greetings, and the messages themselves. A processor controls the compressing, storing, retrieving, forwarding and purging of files. A form of early systems is described in Matthews et al. U.S. Pat. No. 4,371,752 (hereinafter the Matthews '752 Patent), issued in February, 1983, and several related patents. U.S. Pat. No. 4,585,906 (hereinafter the Matthews '906 Patent), issued Apr. 29, 1986 to Gordon H. Matthews et al. The Matthews '906 Patent is a continuation-in-part of the Matthews '752 Patent. U.S. Pat. No. 4,602,129 (hereinafter the Matthews '129 Patent), issued Jul. 22, 1986 to Gordon H. Matthews et al. The Matthews '129 Patent is a continuation-in-part of the '752 Matthews Patent.

The three Matthews Patents each describe a voice mailbox type system using digital storage and programmed control to offer a wide variety of message storage, forwarding and delivery type services.

U.S. Pat. No. 4,625,081, issued Nov. 25, 1986, to Lawrence A. Lotito, et al. This patent describes an automated telephone voice service system which provides automatic recording and editing of voice messages as well as forwarding of recorded voice messages to other accounts and telephone numbers with or without operator assistance.

In all of the foregoing systems voice mail is provided to a single subscriber premises line or, as in the Atwell Patent, to a single subscriber number. A need still exists for an effective and user friendly system for providing personalized calling service features, including actual subscriber identification for voice mail purposes. In particular a need exists for a system for providing personalized features which would facilitate a degree of call control permitting the accomplishment of new functions, including enhanced voice mail and voice mail notification, and which would improve the handling of functions which are now subject to being accomplished only in cumbersome and inconvenient fashions.

DISCLOSURE OF THE INVENTION

The present invention addresses the above noted problems and provides advances over the existing technology by personalizing telecommunication services based on a speech authenticated identification of the not only of the actual subscriber but also of the speakers at both ends of the communication. Offices of a communication network utilize profile data associated with identified persons, rather than profile data associated with a particular telephone number or a particular communication link. In many of the preferred service applications, the network uses a virtual office equipment number assigned to a speaker's profile data to retrieve the data for providing a specific service, reducing or eliminating the need for assignment of additional telephone numbers. The network also provides responding party identification information which is used to determine at least a portion of the processing of the particular call.

Thus, in one aspect the present invention relates to a method of providing service through a communication network. A request to make a call from a predetermined link through the network is detected. The next step in the method is receiving and processing speech signals from a person via the predetermined link. The processing identifies the person making the call as one of a number of subscribers or persons designated as users of services offered through the communication network. An instruction is sent to a switching office of the network instructing that office to utilize profile data

corresponding to the identified subscriber or user for processing of the call. Preferably the profile data is selected at least partially through the use of a virtual office equipment number. This method includes identifying one party to a requested communication service, for example the party making an outgoing call, as one of a plurality of subscribers or designated users. Using a virtual office equipment number, assigned to the identified one user, corresponding profile data is retrieved from storage. A communication network provides the requested communication service over a communication link, based at least in part on the retrieved profile data. As part of the service, a portion of the retrieved profile data is used to direct processing which provides identification of a person responding to the call over another link of the communication network.

Other aspects of the invention relate to a communication network implementing the personalized services, including dual caller and responder specific identification. The system and methodology comprehended by the invention is applicable to both outgoing as well as incoming calls. The preferred implementation of the communication network is an intelligent implementation of a public switched telephone network. The preferred network includes a number of central office switches interconnected by trunk circuits and servicing a substantial number of telephone links. The intelligent network also includes a service control point storing a database of records used in controlling services provided through the central offices. A first signaling network carries signaling messages between the offices as well as signaling messages between the offices and the service control point. A multifunction intelligent peripheral is provided and also may exchange signaling information with the service control point, preferably over a second signaling network.

Another aspect of the invention relates to an improved central office switching system capable of processing a call using profile information selected in response to a virtual equipment number. An office equipment number is 'virtual' where it is assigned to an individual subscriber, instead of to specific network equipment such as a line termination or a specific station.

The switching system includes interface modules coupled to the communication links and a switch providing selective communication connections between the interface modules. An administrative module controls connections provided by the switch. The administrative module includes mass storage containing subscriber profiles, a processor for providing control instructions to the switch, and a signaling interface for signaling communication with at least one external network node. In response to a virtual office equipment number received via the signaling interface, e.g. from a separate peripheral platform as discussed above, the processor retrieves a subscriber profile corresponding to the virtual office equipment number from the mass storage. The processor uses the retrieved profile to process a selective connection through the switch between two of the interface modules.

Advantages of the personal dial tone service should be readily apparent to those skilled in the telecommunications art. For example, in the shared line application, several subscribers can share a single line or communication link as well as a single telephone number. Outgoing call features, however, are personalized to each subscriber. For example, the network can provide each user a different level of service which, according to a preferred embodiment of the invention, may impose restrictions on that user. In addition, the network may direct the performance of a variety of functions both within and without the network. These functions preferably include the identification of the second party to the communication, and the specific nature of the functions are at least in part determined by that identification. The service uses speech based identification.

Additional objects, advantages and novel features of the invention will be set forth in part in the description which follows, and in part will become apparent to those skilled in the art upon examination of the following or may be learned by practice of the invention. The objects and advantages of the invention may be realized and attained by means of the instrumentalities and combinations particularly pointed out in the appended claims.

BRIEF DESCRIPTION OF DRAWINGS

The drawing figures depict the present invention by way of example, not by way of limitations. In the figures, like reference numerals refer to the same or similar elements.

FIG. 1 is a simplified block diagram of an intelligent telephone network that may be used to offer the personalized service of the present invention.

FIG. 2 is a simplified block diagram illustrating the significant functional components of an SSP type central office switching system used in the network of FIG. 1.

FIG. 3 is a simplified block diagram illustrating the significant functional components of an Intelligent Peripheral (IP) used in the network of FIG. 1.

FIG. 4A is a combination signal flow and process flow diagram useful in understanding a specific example of call processing for providing an illustrative personalized service over a shared use line.

FIG. 4B is a combination signal flow and process flow diagram useful in understanding one embodiment of the processing for providing the identity of the actual caller to the destination display as caller ID information.

FIG. 4C is a combination signal flow and process flow diagram useful in understanding another embodiment of the processing for providing the identity of the actual caller to the destination display as caller ID information.

FIG. 5 is a combination signal flow and process flow diagram useful in understanding a specific example of call processing for providing an illustrative personalized service on a dial-up, per call basis.

FIG. 6 is a block diagram depicting an example of one voice mail system suitable for use pursuant to one preferred embodiment of the invention.

BEST MODE FOR CARRYING OUT THE INVENTION

In response to each of several types of service requests, the personalized service of the present invention initially identifies the individual subscriber or user, preferably using a speaker identification/verification procedure. The system then retrieves profile information corresponding to the identified subscriber or user. The communication network processes one or more calls to or from an identified communication link using the individual user's profile data. On an outgoing telephone call from the subscriber or user, for example, the service request may be an off-hook signal, and the network may provide 'dial-tone' type telephone services based on the retrieved profile information. In this example, the network may provide a dial tone signal or a customized prompt and then permit the caller to out-dial a call. Caller identification, calling features and/or additional identification of the responding party functions apply based on the profile information. The network also provides personalized

services on incoming calls based on the identity of the calling party and on data contained in the individual profile of the answering user.

The personalized service may utilize a variety of different networks. For example, the service may be adaptable to Internet based voice communications. The preferred embodiments utilize various implementations of modern telephone networks. To understand the invention, it may be helpful first to consider the architecture and operation of an advanced intelligent network (AIN) type implementation of a public switched telephone network.

FIG. 1 provides a simplified illustration of the preferred intelligent telephone network for implementing the personal dial tone service in accord with the present invention. As shown, the telephone network includes a switched traffic network and a common channel signaling network carrying the control signaling messages for the switched telephone traffic network. In this implementation, the system further includes a secondary signaling network.

The telephone or traffic network (operated by a combination of local carriers and interexchange carriers) includes a number of end office and tandem office type central office switching systems 11. FIG. 1 shows a number of subscriber stations, depicted as telephones 1, connected to a series of central office switches 11_1 to 11_N. In the preferred implementation, the connections to the central office switches 11 utilize telephone lines, and the switches are telephone type switches for providing landline communication. However, it should be recognized that other communication links and other types of switches could be used. Trunk circuits TR carry communication traffic between the central office switches 11.

Each end office type central office switch, such as 11_1 and 11_N, provides switched telephone connections to and from local communication lines or other subscriber links coupled to end users stations or telephone sets 1. For example, the central office 11_1 serves as an end office to provide switched telephone connections to and from local communication lines coupled to end users telephone station sets, such as telephone 1_A, whereas the central office 11_N serves as an end office to provide switched telephone connections to and from local communication lines coupled to end users telephone station sets, such as telephone 1_B.

The typical telephone network also includes one or more tandem switching offices such as office 11_T, providing trunk connections between end offices. As such, the traffic network consists of local communication links and a series of switching offices interconnected by voice grade trunks, only two examples of which are shown at TR in FIG. 1. One set of trunks TR might interconnect the first end office 11_1 to the tandem office 11_T, whereas another set of trunks TR might interconnect the tandem office 11_T to another end office 11_N. Other trunks might directly connect end offices. Although not shown, many offices serve as both end offices and tandem offices for providing different call connections.

FIG. 1 shows connections to the stations 1 via lines, and typically these links are telephone lines (e.g. POTS or ISDN). It will be apparent to those skilled in the art, however, that these links may be other types of communication links, such as wireless links. At least some of the stations have caller ID capability. If the line is an ISDN line, the station may incorporate a display for visually presenting the caller ID information and other signaling related messages. If the link is a typical analog telephone line, the customer premises equipment includes a caller ID terminal, one example of which is shown at 5_B. The terminal 5_B displays at least telephone numbers and preferably displays alphanumeric information to enable displays of callers names.

Although shown as telephones in FIG. 1, the terminal devices or stations 1 can comprise any communication device compatible with the local communication link. Where the link is a standard voice grade telephone line, for example, the terminals could include facsimile devices, modems etc. The processing in accord with the invention, however, relies on identification of the subscriber, preferably by voice based recognition. For this purpose, the terminals preferably include two-way voice communication elements.

The lines and trunks through the central offices 11 carry the communication traffic of the telephone network. The preferred telephone network, however, also includes a common channel interoffice signaling (CCIS) network carrying a variety of signaling messages, principally relating to control of processing of calls through the traffic portion of the network. The CCIS network includes packet data links (shown as dotted lines) connected to appropriately equipped central office switching systems such as offices 11 and a plurality of packet switches, termed Signaling Transfer Points (STPs) 15. To provide redundancy and thus a high degree of reliability, the STPs 15 typically are implemented as mated pairs of STPs. The CCIS network of the telephone system operates in accord with an accepted signaling protocol standard, preferably Signaling System 7 (SS7).

In the preferred embodiment shown in FIG. 1, each central office 11 has at least minimal SS7 signaling capability, which is conventionally referred to as a signaling point (SP) in reference to the SS7 network. As such, the offices can exchanges messages relating to call set-up and tear-down, typically in ISDN-UP format. At least some, and preferably all, of the central office switches 11 are programmed to recognize identified events or points in call (PICs) as advanced intelligent network (AIN) type service triggers. In response to a PIC or trigger, a central office 11 initiates a query through the CCIS signaling network to a control node to either a Service Control Point (SCP) 19 or to a database system, such as a Line Identification Database (LIDB) 21. The SCP 19 provides instructions relating to AIN type services. The LIDB 21 provides subscriber account related information, for calling card billing services or for subscriber name display purposes in an enhanced caller ID application. Those central office switching systems having full AIN trigger and query capability for communication with the SCP and/or the LIDB are referred to as Service Switching Points (SSPs).

The central office switches 11 typically consist of programmable digital switches with CCIS communications capabilities. One example of such a switch is a 5ESS type switch manufactured by AT&T; but other vendors, such as Northern Telecom and Seimens, manufacture comparable digital switches which could serve as the SSPs and SPs. The SSP type implementation of such switches differs from the SP type implementation of such switches in that the SSP switch includes additional software to recognize the full set of AIN triggers and launch appropriate queries. A specific example of an SSP capable switch is discussed in detail later, with regard to FIG. 2.

One key feature of the present invention is that the program controlled switch accepts instructions to load profiles and/or receives profiles over a signaling link. In most cases, these profiles are identified by virtual office equipment numbers. The profiles include a range of information relating to subscribers services, such as service features,

classes of service, individual billing options, and according to a preferred feature of the invention, information relating to restrictions applied to individual users, as well as the performance of functions related to that user.

The above described data signalling network between the SSP type central offices 11 and the SCP 19 is preferred, but other signalling networks could be used. For example, instead of the packet switched type links through one or more STP's, a number of central office switches, an SCP and any other signalling nodes could be linked for data communication by a token ring network. Also, the SSP capability may not always be available at the local office level, and several other implementations might be used to provide the requisite SSP capability. For example, none of the end office switches may have SSP functionality. Instead, each end office would connect through a trunk to a tandem office which has the SSP capability. The SSP tandem then communicates with the SCP via an SS7 type CCIS link, as in the implementation described above. The SSP capable tandem switches are digital switches, such as the 5ESS switch from AT&T; and the non-SSP type end offices might be 1A analog type switches.

The SCP 19 may be a general purpose computer storing a database of call processing information. In the preferred implementation, the SCP 19 actually is an Integrated Service Control Point (ISCP) developed by Bell Atlantic and Bell Communications Research. The ISCP is an integrated system. Among other system components, the ISCP includes a Service Management System (SMS), a Data and Reporting System (DRS) and the actual database also referred to as a Service Control Point (SCP). In this implementation, the SCP maintains a Multi-Services Application Platform (MSAP) database which contains call processing records (CPRs) for processing of calls to and from various subscribers. The ISCP also typically includes a terminal subsystem referred to as a Service Creation Environment or SCE for programming the MSAP database in the SCP for the services subscribed to by each individual customer.

The components of the ISCP are connected by an internal, high-speed data network, such as a token ring network. The internal data network also typically connects to a number of interfaces for communication with external data systems, e.g. for provisioning and maintenance. In the preferred embodiment, one of these interfaces provides communications to and from the SCP 19 via a packet switched data network, such as the TCP/IP network 27.

The SCP may be implemented in a variety of other ways. The SCP may be a general purpose computer running a database application and may be associated with one of the switches. Another alternative is to implement a database of CPRs or the like within an STP (see e.g. Farris et al. U.S. Pat. No. 5,586,177).

The LIDB database 21 is a general purpose computer system having a signalling link interface or connection to a pair of STPs 15. The computer runs a database program to maintain a database of information relating to customer accounts and identifications. For example, a subscriber's entry in the LIDB database might include the subscriber's telephone number, a personal identification number for credit card billing purposes, and the subscriber's name and address.

The preferred telephone network also includes one or more intelligent peripherals (IPs) 23 to provide enhanced announcement and digit collection capabilities and speech recognition. The IP 23 is essentially similar to that disclosed in commonly assigned U.S. U.S. Pat. No. 5,572,583 to Wheeler, Jr. et al. entitled "Advanced Intelligent Network with Intelligent Peripherals Interfaced to the Integrated Services Control Point," and the disclosure of the network and operation of the IP disclosed from that Patent is incorporated herein in its entirety by reference.

The IP 23 may connect to one or more central offices 11. The connections transport both communication traffic and signaling. The connection between a central office 11 and the IP 23 may use a combination of a T1 and a Simplified Message Desk Interface (SMDI) link, but preferably this connection utilizes a primary rate interface (PRI) type ISDN link. Each such connection provides digital transport for a number of two-way voice grade type telephone communications and a channel transporting signaling data messages in both directions between the switch and the IP.

As discussed more later, there are certain circumstances in which the SCP 19 communicates with the IP 23. These communications could utilize an 1129 protocol and go through an SSP type central office 11 and the SS7 network. However, in the preferred embodiment of FIG. 1, the IP 23 and the SCP 19 communicate with each other via a separate second signaling network 27. These communications through network 27 between the IP and the SCP may utilize an 1129+ protocol or a generic data interface (GDI) protocol as discussed in the above incorporated Patent to Wheeler, Jr. et al.

The IP 23 can provide a wide range of call processing functions, such as message playback and digit collection. In the preferred system, the IP also performs speaker identification/verification (SIV) on audio signals received from users. Specifically, the IP 23 used for the personalized service includes a voice authentication module to perform the necessary speaker identification/verification function. The IP 23 also includes storage for subscriber specific template or voice feature information, for use in identifying and authenticating subscribers based on speech.

In the simplest form, the IP 23 serving a subscriber's local area stores the templates and performs the speaker identification/verification. However, in a system serving a large geographic area and providing personal dial tone to a large, roaming subscriber base, the templates may be transferred between SCP/IP pairs, to allow an IP near a subscriber's current location to perform the speaker identification/verification on a particular call. For example, if a remote IP 23_R required a template for a subscriber from the region served by the IP 23, the remote IP 23_R would transmit a template request message through the network 27 to the IP 23. The IP 23 would transmit the requested template back through the network 27 to the remote IP 23_R.

In a network such as shown in FIG. 1, routing typically is based on dialed digit information, profile information regarding the link or station used by the calling party and profile information regarding a line or station in some way associated with the dialed digits. Each exchange is identified by one or more three digit codes. Each such code corresponds to the NXX digits of an NXX-XXXX (seven digit) telephone number or the three digits following the area code digits (NPA) in a ten-digit telephone number. The telephone company also assigns a telephone number to each subscriber line connected to each switch. The assigned telephone number includes the area code and exchange code for the serving central office and four unique digits.

Central office switches utilize office equipment (OE) numbers to identify specific equipment such as physical links or circuit connections. For example, a subscriber's line might terminate on a pair of terminals on the main distri-

bution frame of a switch 11. The switch identifies the terminals, and therefore the particular line, by an OE number assigned to that terminal pair. For a variety of reasons, the operating company may assign different telephone numbers to the one line at the same or different times. For example, a local carrier may change the telephone number because a subscriber sells a house and a new subscriber moves in and receives a new number. However, the OE number for the terminals and thus the line itself remains the same.

On a normal call, an end office type switch will detect an off-hook condition on the line and provide dial tone. The switch identifies the line by its OE number. The office also retrieves profile information corresponding to the OE number and off-hook line. If needed, the profile identifies the currently assigned telephone number. The switch in the end office receives dialed digits and routes the call. The switch may route the call to another line serviced by that switch, or the switch may route the call over trunks and possibly through one or more tandem offices to an office that serves the called party's station or line. The switch terminating a call to a destination will also utilize profile information relating to the destination, for example to forward the call if appropriate, to apply distinctive ringing, etc.

AIN call processing involves a query and response procedure between an SSP capable switching office 11 and a database system, such as the SCP 19. The SSP capable switching offices initiate such processing upon detection of triggering events. At some point during processing of a telephone call, a central office switching system 11 will recognize an event in call processing as a 'Point in Call' (PIC) which triggers a query to the SCP 19. Ultimately, the SCP 19 will return an instruction to the switching system 11 to continue call processing. This type of AIN call processing can utilize a variety of different types of triggers to cause the SSPs 11 to initiate the query and response signaling procedures with the SCP 19. In the presently preferred embodiments discussed below, the personal dial tone service utilizes an off-hook immediate trigger, a dialed number trigger and a terminating attempt trigger (TAT), to facilitate different aspects of the service.

In accord with one aspect of the present invention, before providing dial-tone service, the SSP central office 11 that is serving an outgoing call extends the call to the IP 23 providing the speaker identification/verification (SIV) functionality. In the preferred embodiments, this operation involves AIN type call routing to the IP. The IP 23 prompts the caller and collects identifying information, preferably in the form of speech. The IP analyzes the caller's input to identify the caller as a particular subscriber. If successful, the IP signals the SSP to load profile data for that subscriber into the register assigned to the call in the call store. In most of the preferred service applications, the IP disconnects, and the SSP central office 11 processes the call in accord with the loaded profile information. For example, the central office 11 may now provide actual dial tone or provide a message prompting the caller to dial a destination number. The caller dials digits, and the central office processes the digits to provide the desired outgoing call service, in the normal manner. The IP may stay on the line, to monitor speech and thus caller identity, for some service applications.

The call processing by the central office switch 11 utilizes the loaded subscriber profile information. For example, the profile data may indicate specific procedures for billing the call to this subscriber on some account not specifically linked to the originating telephone station or line. For example, in a college dormitory, the billing information might specify billing of a student's calls to the account of the student's parent(s). Any call restrictions, imposed at the wish of the parents, would be reflected in the profile. The switch would restrict the calling services accordingly, e.g. to limit distance, cumulative cost and/or duration of calls. The dormitory example is to be regarded as merely illustrative of the varied situations to which the system and methodology of the invention is applicable, as will become apparent from following detailed description.

The inventors also envision use of selected subscriber profile information on incoming calls. When a serving central office SSP 11 detects a call to a line having the personalized service, processing hits a terminating attempt trigger (TAT). The SSP interacts with the SCP 19 and routes the call to the IP 23. The IP 23 prompts the caller to identify a desired called party, e.g. one of the students sharing the dormitory line. Menu announcement together with either digit collection or preferably speech recognition processing by the IP 23 facilitates identification of the desired called party from those associated with the line. Based on identification of the called subscriber, the IP 23 signals the SSP switch 11 to load profile data for that subscriber into the register assigned to the call in the call store. In this case, however, the switch 11 uses selectively loaded profile information for terminating the call. The IP disconnects, and the SSP central office 11 processes the call in accord with the loaded profile information.

For example, the central office 11 may provide a distinctive ringing signal corresponding to the identified subscriber. This service enables distinctive ringing for multiple subscribers on one line without assigning each subscriber a separate telephone number. The loaded profile information may specify call forwarding in event of a busy or no-answer condition. This enables routing of the call to the identified subscriber's mailbox, or another alternate destination selected by the subscriber, even though the call did not utilize a unique telephone number uniquely assigned to the called subscriber.

It is a feature of one preferred embodiment of the invention that the menu utilized on an incoming call also includes a so-called 'challenge' wherein the caller is requested to speak his or her name. The profile of the called user which has been installed in response to identification of the user may contain limitations applicable to identified callers. To this end the speech recognition node, preferably the IP, is provided with a previously obtained template to permit identification of such callers. As later described in further detail, the identification of both the called and calling party may entail maintaining a voice connection to the IP. Such a connection may be utilized for either recording the conversation and/or bridging a third party onto the call, such as a parent or other supervisory authority.

The present invention also encompasses a procedure in which a subscriber calls in from a line not specifically designated for personal dial tone service. The network routes the call to the IP 23, and the IP identifies the subscriber and the line from which the subscriber called-in. The subscriber can interact with the IP 23 to have her personal dial tone service associated with that line, either for one call or for some selected period of time. The IP 23 instructs the appropriate central office switch(es) 11 to load profile data associated with the subscriber.

The IP 23 might instruct the end office switch to load the profile data only in the assigned call store register. The switch would use the profile data only for a single call, for example to bill a call from a pay-phone or a hotel room telephone to the subscriber's home account. Alternatively,

the IP 23 might instruct the central office 11 serving the line to the calling station 1 to utilize a virtual office equipment number (OE) and associated profile data for calls to and from that line for some period of time. In this later example, the IP 23 would also instruct the central office 11 serving the line to the subscriber's home station 1 to modify the subscriber's profile to forward calls for the subscriber's telephone number. The modified profile data in the home office 11 would result in forwarding of the subscriber's incoming calls through the office 11 to the selected station 1, for the set period of time.

The present invention relies on the programmable functionality of the central office switches and the enhanced call processing functionalities offered by the IPs. To understand these various functionalities, it may be helpful to review the structure and operation of a program controlled central office and one implementation of an IP. Subsequent description will explain several of the above outlined call processing examples in greater detail.

FIG. 2 is a simplified block diagram of an electronic program controlled switch which may be used as any one of the SSP type central offices 11 in the system of FIG. 1. As illustrated, the switch includes a number of different types of modules. In particular, the illustrated switch includes interface modules 51 (only two of which are shown), a communications module 53 and an administrative module 55.

The interface modules 51 each include a number of interface units 0 to n. The interface units terminate lines from subscribers' stations, trunks, T1 carrier facilities, etc. Each such termination is identified by an OE number. Where the interfaced circuit is analog, for example a subscriber loop, the interface unit will provide analog to digital conversion and digital to analog conversion. Alternatively, the lines or trunks may use digital protocols such as T1 or ISDN. Each interface module 51 also includes a digital service unit (not shown) which is used to generate call progress tones and receive and detect dialed digits in pulse code or dual-tone multifrequency form.

In the illustrated embodiment, the unit 0 of the interface module 51' provides an interface for the signaling and communication links to the IP 23. In this implementation, the links preferably consist of one or more ISDN PRI circuits each of which carries 23 bearer (B) channels for communication traffic and one data (D) channel for signaling data.

Each interface module 51 includes, in addition to the noted interface units, a duplex microprocessor based module controller and a duplex time slot interchange, referred to as a TSI in the drawing. Digital words representative of voice information are transferred in two directions between interface units via the time slot interchange (intramodule call connections) or transmitted in two directions through the network control and timing links to the time multiplexed switch 57 and thence to another interface module (intermodule call connection).

The communication module 53 includes the time multiplexed switch 57 and a message switch 59. The time multiplexed switch 57 provides time division transfer of digital voice data packets between voice channels of the interface modules 51 and transfers signaling data messages between the interface modules. The switch 57 together with the TSIs of the interface modules form the overall switch fabric for selectively connecting the interface units in call connections.

The message switch 59 interfaces the administrative module 55 to the time multiplexed switch 57, so as to provide a route through the time multiplexed switch permitting two-way transfer of control related messages between the interface modules 51 and the administrative module 55. In addition, the message switch 59 terminates special data links, for example a link for receiving a synchronization carrier used to maintain digital synchronism.

The administrative module 55 provides high level control of all call processing operations of the switch 11. The administrative module 55 includes an administrative module processor 61, which is a computer equipped with disc storage 63, for overall control of CO operations. The administrative module processor 61 communicates with the interface modules 51 through the communication module 53. The administrative module 55 may include one or more input/output processors (not shown) providing interfaces to terminal devices for technicians and data links to operations systems for traffic, billing, maintenance data, etc.

A CCIS terminal 73 and an associated data unit 71 provide an SS7 signalling link between the administrative module processor 61 and one of the STPs 15 (see FIG. 1). Although only one such link is shown, preferably there are a plurality of such links providing redundant connections to both STPs of a mated pair and providing sufficient capacity to carry all necessary signaling to and from the particular office 11. The SS7 signaling through the terminal 73, the data unit 71 and the STPs provides two-way signaling data transport for call set-up related messages to and from other offices. These call set-up related messages typically utilize the ISDN-UP (ISDN-users part) protocol portion of SS7. The SS7 signaling through the terminal 73, the data unit 71 and the STPs also provides two-way signaling data transport for communications between the office 11 and database systems or the like, such as the SCP 19. The communications between the office 11 and the database systems or the like utilize the TCAP (transactions capabilities applications part) protocol portion of SS7.

As illustrated in FIG. 2, the administrative module 55 also includes a call store 67 and a program store 69. Although shown as separate elements for convenience, these are typically implemented as memory elements within the computer serving as the administrative module processor 61. The program store 69 stores program instructions which direct operations of the computer serving as the administrative module processor 61.

For each call in progress, a register assigned within the call store 67 stores translation and user profile information retrieved from disc storage 63 together with routing information and any temporary information needed for processing the call. For example, for a residential customer initiating a call, the call store 67 would receive and store line identification and outgoing call billing information corresponding to an off-hook line initiating a call. For the personal dial-tone service, the assigned register in the call store 67 will receive and store different profile data depending on the particular subscriber associated with any given call. A register in the call store is assigned and receives profile data from the disc memory both for originating subscribers on outgoing calls and for terminating subscribers on incoming calls.

A variety of adjunct processor systems known in the telephone industry can be used as the IP 23. The critical requirements are that the IP system process multiple calls and perform the subscriber identification functions, preferably by speaker identification and authentication. FIG. 3 is a functional diagram illustration of an IP 23 for performing the subscriber identification functions, possibly by dialed digit input and preferably by analysis and recognition of speech.

The preferred IP architecture utilizes separate modules for different types of services or functions, for example, one or two Direct Talk type voice server modules 231A, 231B for interfacing ISDN PRI trunks to the SSP central office(s) 11. Separate modules 233, 235 perform voice authentication and speech recognition. The IP 23 includes a variety of additional modules for specific types of services, such as a server module 237 for fax mail, and another server 239 for voice mail services. The various modules communicate with one another via an internal data communication system or bus 240, which may be an Ethernet type local area network.

Each Direct Talk module 231A or 231B comprises a general purpose computer, such as an IBM RS-6000, having digital voice processing cards for sending and receiving speech and other audio frequency signals, such as IBM D-talk 600 cards. Each voice processing card connects to a voice server card which provides the actual interface to T1 or primary rate interface ISDN trunks to the switching office. In the PRI implementation, the Direct Talk computer also includes a signaling card, providing two-way signaling communication over the D-channel of the PRI link. Each Direct Talk computer also includes an interface card for providing two-way communications over the internal data communications system 240.

The voice processing cards in the Direct Talk modules 231A, 231B provide voice message transmission and dialed digit collection capabilities. The modules 231A, 231B also perform the necessary line interface functions for communications to and from those servers which do not incorporate actual line interfaces. For example, for facsimile mail, a Direct Talk module 231 connected to a call would demodulate incoming data and convert the data to a digital format compatible with the internal data communication network 240. The data would then be transferred over network 240 to the fax server 237. For outgoing facsimile transmission, the server 237 would transfer the data to one of the Direct Talk modules over the network 240. The Direct Talk module 231 would reformat and/or modulate the data as appropriate for transmission over the ISDN link to the switch 11.

The Direct Talk modules provide a similar interface function for the other servers, such as the voice mail server 239, the speech recognition module 235 and the voice authentication module 233. For incoming speech signals, the Direct Talk module connected to a call receives digital speech signals in the standard pulse code modulation format carried on a B-channel of an ISDN link. The Direct Talk module reformats the speech data and transmits that data over the internal network 240 to the server or module performing the appropriate function, for example to the authentication module 233 for analysis and comparison of features to stored templates or feature data for known subscribers.

In the outgoing direction, the currently connected Direct Talk module may play an announcement from memory, e.g. to prompt a caller to say their name. Alternatively, the Direct Talk module may receive digitized speech over the network 240 from one of the other modules, such as a stored message retrieved from voice mail server 239. The Direct Talk module reformats the speech signal as needed for transmission over the ISDN B-channel to the caller.

The illustrated IP also includes a communication server 243. The communication server 243 connects between the data communication system 240 and a router 241, which provides communications access to the TCP/IP network 27 that serves as the second signaling communication system. The communication server 243 controls communications

between the modules within the IP 23 and the second signaling communication system. The server 243 and the router 241 facilitate communication between the elements of the IP 23 and the SCP 19. The IP may also use this communication system to communicate with other IP's, for example to send subscriber voice template information to the remote IP 23_R (FIG. 1) or to receive such information from that IP or some other network node.

The personalized service relies on the voice authentication module 233 to perform the necessary speaker identification/verification function. For the identification and authentication of subscribers or users, the voice authentication module 233 within the IP 23 stores a template or other feature or voice pattern information for each person who has the personalized service in the area that the IP services. For example, if the subscriber utilizes the personal dial tone service from a particular line, such as a shared line in a dormitory or the like, the IP stores the subscriber's voice pattern information in a file associated with the office equipment (OE) number of the particular line. If the IP 23 serving a call does not store the template or feature data for a particular subscriber, the IP 23 may obtain subscriber identification by dialed digit input and then obtain a copy of the template or feature data from a remote IP 23_R via communication through the TCP/IP network 27, in order to authenticate the subscriber's identity.

Using current technology, a new subscriber or user would get on line with the IP serving that subscriber and 'train' that IP by speaking certain phrases. From the received audio signals representing those phrases, the IP would store templates or other pattern information for use in identifying and/or verifying that a caller is the particular subscriber.

During actual call processing, the voice authentication module 233 receives speech information from the caller. The voice authentication module 233 compares the received information to its stored template or feature data to identify a calling party as a particular subscriber.

In the case of speech recognition applied to incoming calls, the IP is trained in a different manner. Current speech recognition technology permits recognition with a reasonable degree of certitude based on training from a limited sample of recorded speech of a subject. In situations where the target of the speech recognition is not such as may participate in the cooperative manner of subscribers, recorded samples of prior telephone speech may be used with available recognition facilities of a more sophisticated nature.

In such situations the present invention also relies on the speech recognition capability of the module 235, particularly in processing of incoming calls in certain situations. The speech recognition module 235 enables the IP to analyze incoming audio information to recognize vocabulary words. The IP 23 interprets the spoken words and phrases to determine subsequent action. For example, the IP might recognize the caller speaking the name of a called subscriber and use the subscriber identification to instruct the terminating central office to control the call in accord with that subscriber's profile.

The preferred routing of the calls in accord with the invention utilizes AIN type call processing. To understand the call processing, it may be helpful to consider several specific examples in more detail.

In a first example, consider an outgoing call from the station 1_A to the station 1_B. Assume per call assignment of profile data to the originating line, for personal dial tone service on each outgoing call. FIG. 4 provides a simplified flow diagram of the signal flow and processing for such an outgoing call.

Assume use of a standard telephone for purposes of this example. The person lifts the handset creating an off-hook state in the telephone 1_A, and a corresponding signal or change in state on the line to the central office 11 (step S1). In this call flow, the off-hook signal is a type of service request, i.e. a request to make an outgoing call. The serving central office 11_1 detects the off-hook and commences its call processing. Specifically, the central office assigns a register in the call store 67 to this call and loads profile information associated with the off-hook line from the disc storage 63 into the assigned register. In this case, the central office 11_1 is an SSP capable office, and the loaded profile data indicates an off-hook immediate trigger set against the particular line. The serving SSP type office 11_1 therefore detects this off-hook PIC as an AIN trigger (step S2).

In response to the off-hook and the off-hook trigger set in the subscriber's profile, the SSP type central office switch 11_1 launches a query to the SCP 19 (step S3). Specifically, the SSP 11_1 creates a TCAP query message containing relevant information, such as the office equipment (OE) number assigned to the off-hook line, and transmits that query over an SS7 link to one of the STPs 15. The query includes a destination point code and/or a global title translation addressing the message to the SCP 19, and the STP 15 relays the query message over the appropriate link to the SCP 19. The query from the SSP central office 11_1 identifies the caller's line by its associated office equipment (OE) number and possibly by a single telephone number associated with the off-hook line.

In response to a query, the SCP 19 accesses its a database, typically, the MSAP database set up in the ISCP, to determine how to process the particular call. The SCP 19 identifies an access key in the query and uses the key to retrieve the appropriate record from the database. In this case, the query indicates an off-hook trigger as the trigger event, therefore the SCP 19 uses the calling party office equipment (OE) number as the access key. The SCP 19 retrieves a call processing record (CPR) corresponding to the office equipment (OE) number associated with the off-hook line and proceeds in accord with that CPR (step S4).

For the present example of the personal dial tone service, the CPR will provide information necessary for routing the call to some node of the network that will perform speaker identification/verification (SIV). In the preferred embodiment, the SIV is a function performed by an Intelligent Peripheral (IP), therefore the CPR provides information for routing the call to the nearest available IP having the SIV capability.

Based on the CPR, the SCP 19 formulates a response message instructing the SSP central office 11_1 serving the customer to route the call. In this case, the message includes information, e.g. a office equipment (OE) number or telephone number, used for routing a call to the identified IP 23. The SCP 19 formulates a TCAP message in SS7 format, with the destination point code identifying the SSP office 11_1. The SCP 19 transmits the TCAP response message back over the SS7 link to the STP 15, and the STP 15 in turn routes the TCAP message to the SSP central office 11_1 (see step S5) The SSP type switch in the central office 11_1 uses the routing information to connect the call to one of the lines or channels to the IP 23. A two-way voice grade call connection now extends between the calling station 1_A and the IP 23 (step S6). In the present example, the switch actually connects the off-hook line to the line to the IP before providing dial tone.

As noted above, the communication link to the IP 23 provides both line connections and signaling, preferably over a primary rate interface (PRI) type ISDN link. When the central office 11_1 extends the call from the calling party's line to a line circuit (over a B channel) to the IP 23, the switch in that office also provides call related data over the signaling link (D channel for ISDN). The call related data, for example, includes the office equipment (OE) number normally associated with the off-hook line and possibly the telephone number for that line.

In response to the incoming call, the IP 23 will seize the line, and it will launch its own query to the SCP 19 (step S7). In the preferred network illustrated in FIG. 1, the IP 23 and the SCP 19 communicate with each other via a separate second signalling network 27, for example utilizing either an 1129+ protocol or a generic data interface (GDI) protocol as discussed in U.S. Pat. No. 5,572,583 to Wheeler, Jr. et al. The query from the IP 23 again identifies the caller's line by at least its associated office equipment (OE) number.

In response to the query from the IP 23, the SCP 19 again accesses the appropriate CPR (step S8) and provides a responsive instruction back through the network 27 to the IP 23 (step S9). Although the IP 23 could passively monitor any speech that the user might utter, the preferred implementation utilizes a 'Challenge Phase' to prompt the user to input specific identifying information. In this case, the instruction causes the IP 23 to provide a prompt message over the connection to the caller (step S10). Here, the signal to the caller may be a standard dial tone or any other appropriate audio signal. Preferably, the instruction from the SCP 19 causes the IP 23 to provide an audio announcement prompting the caller to speak personal information. In one preferred example, in step S10 the IP plays an audio prompt message asking the caller, 'Please say your full name'. The process may ask for any appropriate identifying information.

The signal received by the IP 23 goes over the lines and through the central office switch(es) for presentation via the off-hook telephone 1_A to the calling party. In response, the caller will speak identifying information into their off-hook telephone, and the network will transport the audio signal to the IP 23 (step S11).

As noted above, an IP 23 can provide a wide range of call processing functions, such as message playback and digit collection. In the preferred system, the IP also performs speaker identification/verification (SIV) on the audio signal received from the off-hook telephone in step S11. When the IP 23 receives speech input information during actual call processing, for this service example, the IP analyzes the speech to extract certain characteristic information (step S12).

The IP 23 stores a template or other voice pattern information for each person who has the personalized service in the area that the IP normally services. If the IP 23 does not store the particular template or feature information it needs to process a call, the IP 23 can communicate with a remote IP 23R to obtain that information. In the present shared line example, the IP 23 will store template or feature data for each subscriber associated with the particular off-hook line.

When the IP 23 receives input speech and extracts the characteristic information during actual call processing, the IP compares the extracted speech information to stored pattern information, to identity and authenticate the particular caller. In the present example, the voice authentication module 233 in the IP 23 compares the extracted speech information to the stored template or feature data for each subscriber associated with the particular off-hook line.

In step S13, the IP 23 determines if the information extracted from the speech input matches any of the stored

template data feature data for an identifiable subscriber (within some threshold level of certainty). If there is a match, the IP now knows the identity of the calling subscriber. Based on the identification of the calling subscriber, the IP 23 selects a virtual office equipment (OE) number from storage that corresponds to the subscriber.

The IP 23 formulates a D-channel signaling message containing the virtual office equipment (OE) number together with an instruction to load that OE number into the register assigned to the call in place of the OE number of the off-hook line. The IP 23 supplies the message to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link (step S14). In response, the administrative module processor 61 rewrites the OE number in the register assigned to the call using the OE number received from the IP 23.

Upon rewriting the OE number in the register, the administrative module processor 61 of central office switch 11_1 also reloads the profile information in the register (step S15). Specifically, the administrative module processor 61 retrieves profile information associated with the virtual office equipment (OE) number from the disc storage 63 into the register. As such, the profile information in the assigned register in the call store 67 now corresponds to the identified subscriber, rather than to the off-hook line.

The profile information provides a wide range of data relating to the subscriber's services. The profile data provides necessary billing information, enabling billing from the call to this particular subscriber. The profile also defines various service features available to this subscriber on outgoing calls, such as three-way calling. The profile may define a class of calling service available to the subscriber. In the dormitory example, the caller may be allowed a set dollar amount for long distance calls per month (e.g. $50.00). The profile data will indicate the remaining amount at the time of the call and will cause the switch to interrupt service when the available amount is exhausted. Other class of service restrictions might enable long distance calls only if collect and/or only if calling one or two specified numbers (e.g. only to the parents' house). The class of service might enable only long distance calls within a region or country but not international calls.

In the presently preferred implementation, when the central office switch 11_1 reloads the profile, the central office disconnects the link to the IP 23 and connects tone receivers to the caller's line. Optionally, the central office 11_1 may provide a 'dial tone' or other message over the line (step S16) The caller now dials digits in the normal manner (step S17), and the switch in the central office 11_1 loads the dialed digits into the assigned register within the call store 67. The central office 11_1 utilizes the dialed digits and the subscriber's profile data to process the call (S18). For example, if the dialed digits represent a call within the subscriber's permitted class of service, the switch completes the call to the destination station 1_B using the dialed digits in the normal manner. If the profile data requires a particular billing treatment, e.g. to bill a long distance call to the subscriber, the switch makes the appropriate record and forwards the record to the exchange carrier company's accounting office equipment. In accord with another aspect of the invention, the network provides caller ID data naming the identified subscriber to the destination station.

The processing to complete the call, performed in step S18, actually involves a sequence of steps. Of particular note, some of these steps facilitate delivery of caller ID information to the destination station. The present invention involves delivering caller ID information which corresponds to the identified subscriber, preferably the subscriber's name, rather than simply the number of the line or station from which the subscriber initiates the call. Two processing methodologies are envisioned for providing this calling subscriber ID feature, one involving access to name information in a central database such as LIDB and the other relying on name data from the subscriber's profile.

FIG. 4B is a simplified process and signal flow diagram, illustrating the call completion operations, including caller ID display using data from the profile. The network performs the steps depicted in FIG. 4B after identification of the subscriber, preferably based on speaker identification/verification (SIV). As discussed earlier, the IP 23 supplies the signaling message containing the virtual office equipment (OE) number and the instruction to load that OE number into the assigned register to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link (step S14). In response, the administrative module processor 61 rewrites the OE number in the register and reloads the profile information in the register (step S15).

The central office 11_1 provides dial tone or the like over the line (step S16), the caller dials digits corresponding to the desired destination (step S17), and the switch in the central office 11_1 begins is processing to route the call through the network. Initially, the central office 11_1 uses the dialed number to initiate a CCIS communication with the exchange serving the intended destination, in the example the terminating central office 11_N.

Specifically, the subscriber's serving central office 11_1 generates an Initial Address Message (IAM) for transmission to the terminating central office 11_1 (S181). The IAM message includes the SS7 destination point code (DPC) of the terminating central office 11_N and the SS7 origination point code (OPC) of the customer's serving-end central office 11_1, for addressing purposes. The payload portion of the IAM message includes the called and calling numbers. In accord with the invention, the originating end office 11_1 reads name data from the identified subscriber's profile, currently loaded in the assigned register, and places that data in additional field of the IAM message or in an accompanying information message addressed in the same manner as the IAM message. The originating central office transmits the IAM message and possibly an accompanying information message through the CCIS network to the distant terminating office 11_N (step S181).

When the terminating office 11_N receives the IAM message, the administrative module processor for that office retrieves the customer profile for the number in the destination number field of that message (e.g. the number for the telephone 1_B) from its mass storage system and loads that profile into one of its call store registers. If the called party has an enhanced caller ID service, with name display, the terminating central office 11_N would normally recognize the attempt to complete to that party's number message as a terminating attempt trigger (TAT) type point in call (PIC) to trigger access to the LIDB database for name information. However, in this embodiment of the invention, the terminating end office detects the receipt of the subscriber's name data with the IAM message, therefore the administrative module processor in that office overrides the trigger.

The terminating central office switching system 11_N transmits an Address Complete Message (ACM) back to the central office 11_1 and if the called line is available applies ringing signal to the called party's line (S182). The ACM includes a variety of information, including a calling party status indicator, e.g. line free or busy. If the line is not busy,

the end office 13 rings the station Y corresponding to the dialed digits 703-333-5678, and generates the appropriate indicator in the Address Complete Message (ACM) to indicate that it received the request for a call and that the number is not busy. The ACM message is sent back by simply reversing the point codes from the IAM message. Now the destination point code (DPC) is the point code of the central office 11, and the origination point code (OPC) is the point code of the central office 13. In response to the ACM message, if the called line is available, the originating central office 11 applies a ringback tone signal to the line to the calling station 1_A (S183).

As part of its operations to ring the called telephone station, the terminating central office 11_N transmits a caller ID signal over the line. If the called party has ISDN service or the like, the switch sends a signaling message along with the ringing signal. If the called party has analog telephone service, the switch 11_N transmits a caller ID message (step S184) as frequency shift keyed (FSK) data inserted in the silent interval between the first ringing signal (step S182) and the second ringing signal (S185) applied to the called party's line.

In accord with the invention, the caller ID message applied to the called party's line includes the telephone number associated with the calling station 1_A and at least some additional data specific to the identified subscriber. If the called party has enhanced caller ID for displaying name data, the ISDN telephone or the caller ID terminal 5_B receives the number and the name data received with the IAM message in step S181. The caller ID terminal 5_B or a display device in the ISDN telephone displays the received number and name information, identifying the actual calling party, for review before the called party chooses to answer the call.

If the called party subscribes only to normal caller ID, the end office switch 11_N can transmit only a limited amount of information. For this purpose, the switch will select and transmit one or two characters from the subscriber identification data along with the telephone number. For example, if four persons normally call from the particular originating telephone station or line, the data sent to the terminating central office 11_N might include a letter or number identifying each subscriber. The switch 11_N would transmit that letter or number with the telephone number in the caller ID message for display.

If someone answers the telephone station 1_B, the terminating central office switching system 11_N detects an off-hook condition (S13) and sends an Answer Message (ANM) back to the originating central office 11_1 through one or more of the STPs 15. The ANM message indicates that the called telephone 1_B was picked up. Also, at that time the actual telephone traffic trunk circuit is connected together between the central offices 11_1 and 11_N. The central offices 11 connect the lines to the stations to the respective ends of the trunk circuit, to complete the voice path. At this point, actual voice communication is established between the calling station 1_A and the called station 1_B. Communication continues until one or both parties hang up, at which time, all of the switched connections are torn down.

FIG. 4C is a simplified process and signal flow diagram, illustrating the call completion operations including, caller ID display involving access to name information in a central LIDB database. The network performs the steps depicted in FIG. 4C after identification of the subscriber, preferably based on speaker identification/verification (SIV). As in the example of FIG. 4B, the central office switch 11_1 receives an instruction containing the subscriber's virtual office equipment (OE) number (step S14), loads the corresponding profile information in the register (step S15) and sends dial tone or the like over the line (step S16). The subscriber dials digits corresponding to the desired destination (step S17), and the switch in the central office 11_1 transmits an IAM message through the interoffice signaling network to the terminating central office 11_N. The information sent in or with the IAM message in step S191, however, is different than in the earlier example.

In this embodiment, the originating end office 11_1 reads a short code identifier from the identified subscriber's profile, currently loaded in the assigned register, and places that identifier in additional field of the IAM message or in an accompanying information message addressed in the same manner as the IAM message. For example, if the network provides personal dial tone service to four identified persons associated with the originating telephone 11_1, the short code might comprise a number from zero to three or letters such as A, B, C and D, identified by the state of two bits in the IAM or accompanying information message.

As in the earlier example, the originating end office 11_1 addresses and transmits the IAM message with the specific subscriber identifier code through the SS7 signaling network for receipt by the terminating office 11_N. If the called party has only normal caller ID service, then the terminating office 11_N would transmit a normal caller ID message to the destination, with the identifier appended to the calling party telephone number as an extra digit or character. If the called party often receives calls from this subscriber, even the limited subscriber specific identification provided by the code will enable the called party to recognize that the current call is from the identified subscriber.

FIG. 4B depicts the processing steps, beginning in step S192, for processing a call to a called customer having the enhanced caller ID service for name and number display. In such a case, when the terminating office 11_N, the administrative module processor in that office loads the profile for the called subscriber's telephone number into a register in the call store assigned to this call. Of particular note, because the called customer has the enhanced name and number type caller ID service, the customer profile record establishes a terminating attempt trigger (TAT) against the that customer's telephone number.

At this point, the terminating office 11_N recognizes the called party telephone number in the destination number field of the IAM message as a terminating attempt trigger (TAT) type point in call or PIC (step S192). In response to this PIC, the terminating office 11_N launches a second query message through one or more of the STP(s) 15 to the LIDB database 21 (step S193). The query message includes both the telephone number associated with the calling station 1_A or its telephone line as well as the code identifying the specific subscriber making that call.

The LIDB database 21 uses the calling party telephone number and the code identifying the specific subscriber, received in the query, to retrieve that one subscriber's account file record from the database (step S194). The query also indicates the cause of the query, i.e. the TAT triggering event. From this information, the LIDB database recognizes that the query is a request for name information. The database 21 therefore reads up to 15 characters of name data from the subscriber's account file. The LIDB database 21 compiles a TCAP call control message including the name data and returns that call control message to the terminating central office 11_N via the SS7 network.

The terminating central office switching system 11_N receives the call control message from the LIDB database 21.

To provide the caller ID service in this embodiment, the terminating end office 11_N combines the name data from the call control message together with the calling party number as two caller ID messages. The end office 11_N then signals the originating office 11_1 and initiates ringing of the called party's line, as discussed in more detail below.

Assuming for this discussion that the called line is available, the terminating central office switching system 11_N transmits an Address Complete Message (ACM) indicating availability back to the central office 11_1 and applies ringing signal to the called party's line (step S182). In response to the ACM message, if the called line is available, the originating central office 11 applies a ringback tone signal to the line to the calling station 1_A (S183).

As part of its operations to ring the called telephone station, the terminating central office 11_N transmits a caller ID signal over the line. If the called party has ISDN service or the like, the switch sends the caller ID signaling messages along with the ringing signal. If the called party has analog telephone service, the switch 11_N transmits the caller ID messages sequentially over the line (step S184) as frequency shift keyed (FSK) data inserted in the silent interval between the first ringing signal (step S182) and the second ringing signal (S185) applied to the called party's line. As in the earlier example, the display provides the telephone number associated with the calling station 1_A as well as the name data for the specifically identified calling subscriber.

In the shared line example, each person normally expected to use the line to station 1_A is a different subscriber to the personal dial tone service. As the subscribers make outgoing calls, they each receive their own individualized service over the line on each separate call, in precisely the manner described above relative to steps S1 to S18 and the personal caller ID as described above relative to FIGS. 4B and 4C. For example, each subscriber may receive a different level of calling privileges and/or class of service based on their ability and/or desire to pay for telephone services. Also, the called party receives caller ID information including both the origination telephone number and the name or other identifying information associated specifically with the calling subscriber.

Returning to step S13 in FIG. 4A, the extracted information characterizing the input speech signals may not match any of the templates or feature data used by the IP 23. In this event, the process flows to step S19. The IP will count the number of tries or attempts to identify the subscriber and permit some maximum number of failed attempts (N). Assume, for example, that the software allows only two identification attempts on one call (N=2). On the first failure, the number of tries is less than N, therefore processing returns to step S10, and the IP 23 again transmits the prompt for speech input. The caller again speaks the requested input information (S11), and the authentication module 233 again analyzes the input information (S12). If the second input adequately matches a stored subscriber's information in step S13, the processing flows through steps S14 to S18 to complete the call as described above.

However, if the extracted speech information does not match a stored subscriber template or feature data, processing again flows to step S19. If the number of tries now corresponds to the limit N, for example on the second failed attempt, the processing branches to step S20. The IP 23 may now transmit a message indicating denial of service, although this is optional. If provided, the message states that only a limited class of service is available in view of the problems in recognizing the caller as a known subscriber.

The IP 23 formulates a D-channel signaling message instructing the central office switch 11_1 to process the call in accord with default conditions and transmits that instruction to the central office switch (step S21). The instruction could include a default OE number corresponding to a default profile, or the message could instruct the switch to proceed using the OE and profile data for the off-hook line itself. The IP 23 supplies the message to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link (step S21). The administrative module processor 61 resumes call processing using the appropriate default OE and profile.

In the preferred embodiment, the switch provides a normal dial tone (S22), collects dialed digits from the caller (S23) and processes the call (S24). However, the default profile provides only some limited class of service, for example only emergency 911 service or 911 service plus flat rate local calling. The default call processing provides no additional information from the profile corresponding to any particular subscriber, therefore the network processes the call as a normal call for caller ID purposes. The caller ID service will provide only the telephone number to callers having normal caller ID, and the network will access LIDB database 21 to provide name information if any associated strictly with the telephone number, essentially in the manner that the network provides such services when there is no personalized dial tone service involved.

In the above example, the network disconnected the IP 23 after identifying the subscriber and providing the subscriber's virtual OE number to the serving central office 11. For some applications of the personal dial tone service, the central office 11 would maintain a bridged connection of the IP 23 on the line, to enable the IP to monitor the call. For example, in a prisoner telephone service, each prisoner would have only limited telephone rights as specified in each prisoner's profile data. To prevent one prisoner from selling their telephone service rights to another prisoner, the IP 23 would periodically or constantly monitor the outgoing speech signals from the prison line. The voice authentication module 233 would initially identify the prisoner subscriber as discussed above, and would periodically recheck to authenticate the identity of the party using the prison line. If the voice authentication module detects some other party using the line or did not detect the identified subscriber's speech for some predefined time interval, the IP 23 would instruct the serving central office switch 11 to disconnect the call. The IP 23 may send messages to the switch or to other network elements to initiate additional action, such as profile modification to further limit a particular prisoner's telephone privileges and/or to notify prison authorities of misuse of telephone privileges.

If the called party subscribes only to normal caller ID, the end office switch 11_N can transmit only a limited amount of information. For this purpose, the switch will select and transmit one or two characters from the subscriber identification data along with the telephone number. For example, if four persons normally call from the particular originating telephone station or line, the data sent to the terminating central office 11_N might include a letter or number identifying each subscriber. The switch 11_N would transmit that letter or number with the telephone number in the caller ID message for display.

If someone answers the telephone station 1_B, the terminating central office switching system 11_N detects an off-

hook condition (S13) and sends an Answer Message (ANM) back to the originating central office 11_1 through one or more of the STPs 15. The ANM message indicates that the called telephone 1_B was picked up. Also, at that time the actual telephone traffic trunk circuit is connected together between the central offices 11_1 and 11_N. The central offices 11 connect the lines to the stations to the respective ends of the trunk circuit, to complete the voice path. At this point, actual voice communication is established between the calling station 1_A and the called station 1_B. Communication continues until one or both parties hang up, at which time, all of the switched connections are torn down.

FIG. 4C is a simplified process and signal flow diagram, illustrating the call completion operations including, caller ID display involving access to name information in a central LIDB database. The network performs the steps depicted in FIG. 4C after identification of the subscriber, preferably based on speaker identification/verification (SIV) As in the example of FIG. 4B, the central office switch 11_1 receives an instruction containing the subscriber's virtual office equipment (OE) number (step S14), loads the corresponding profile information in the register (step S15) and sends dial tone or the like over the line (step S16). The subscriber dials digits corresponding to the desired destination (step S17), and the switch in the central office 11_1 transmits an IAM message through the interoffice signaling network to the terminating central office 11_N. The information sent in or with the IAM message in step S191, however, is different than in the earlier example.

In this embodiment, the originating end office 11_1 reads a short code identifier from the identified subscriber's profile, currently loaded in the assigned register, and places that identifier in additional field of the IAM message or in an accompanying information message addressed in the same manner as the IAM message. For example, if the network provides personal dial tone service to four identified persons associated with the originating telephone 11_1, the short code might comprise a number from zero to three or letters such as A, B, C and D, identified by the state of two bits in the IAM or accompanying information message.

As in the earlier example, the originating end office 11_1 addresses and transmits the IAM message with the specific subscriber identifier code through the SS7 signaling network for receipt by the terminating office 11_N. If the called party has only normal caller ID service, then the terminating office 11_N would transmit a normal caller ID message to the destination, with the identifier appended to the calling party telephone number as an extra digit or character. If the called party often receives calls from this subscriber, even the limited subscriber specific identification provided by the code will enable the called party to recognize that the current call is from the identified subscriber.

FIG. 4B depicts the processing steps, beginning in step S192, for processing a call to a called customer having the enhanced caller ID service for name and number display. In such a case, when the terminating office 11_N, the administrative module processor in that office loads the profile for the called subscriber's telephone number into a register in the call store assigned to this call. Of particular note, because the called customer has the enhanced name and number type caller ID service, the customer profile record establishes a terminating attempt trigger (TAT) against the that customer's telephone number.

At this point, the terminating office 11_N recognizes the called party telephone number in the destination number field of the IAM message as a terminating attempt trigger (TAT) type point in call or PIC (step S192). In response to this PIC, the terminating office 11_N launches a second query message through one or more of the STP(s) 15 to the LIDB database 21 (step S193). The query message includes both the telephone number associated with the calling station 1_A or its telephone line as well as the code identifying the specific subscriber making that call.

The LIDB database 21 uses the calling party telephone number and the code identifying the specific subscriber, received in the query, to retrieve that one subscriber's account file record from the database (step S194). The query also indicates the cause of the query, i.e. the TAT triggering event. From this information, the LIDB database recognizes that the query is a request for name information. The database 21 therefore reads up to 15 characters of name data from the subscriber's account file. The LIDB database 21 compiles a TCAP call control message including the name data and returns that call control message to the terminating central office 11_N via the SS7 network.

The terminating central office switching system 11_N receives the call control message from the LIDB database 21.

To provide the caller ID service in this embodiment, the terminating end office 11_N combines the name data from the call control message together with the calling party number as two caller ID messages. The end office 11_N then signals the originating office 11_1 and initiates ringing of the called party's line, as discussed in more detail below.

Assuming for this discussion that the called line is available, the terminating central office switching system 11_N transmits an Address Complete Message (ACM) indicating availability back to the central office 11_1 and applies ringing signal to the called party's line (step S182). In response to the ACM message, if the called line is available, the originating central office 11 applies a ringback tone signal to the line to the calling station 1_A (S183).

As part of its operations to ring the called telephone station, the terminating central office 11_N transmits a caller ID signal over the line. If the called party has ISDN service or the like, the switch sends the caller ID signaling messages along with the ringing signal. If the called party has analog telephone service, the switch 11_N transmits the caller ID messages sequentially over the line (step S184) as frequency shift keyed (FSK) data inserted in the silent interval between the first ringing signal (step S182) and the second ringing signal (S185) applied to the called party's line. As in the earlier example, the display provides the telephone number associated with the calling station 1_A as well as the name data for the specifically identified calling subscriber.

In the shared line example, each person normally expected to use the line to station 1_A is a different subscriber to the personal dial tone service. As the subscribers make outgoing calls, they each receive their own individualized service over the line on each separate call, in precisely the manner described above relative to steps S1 to S18 and the personal caller ID as described above relative to FIGS. 4B and 4C. For example, each subscriber may receive a different level of calling privileges and/or class of service based on their ability and/or desire to pay for telephone services. Also, the called party receives caller ID information including both the origination telephone number and the name or other identifying information associated specifically with the calling subscriber.

Returning to step S13 in FIG. 4A, the extracted information characterizing the input speech signals may not match any of the templates or feature data used by the IP 23. In this

event, the process flows to step S19. The IP will count the number of tries or attempts to identify the subscriber and permit some maximum number of failed attempts (N). Assume, for example, that the software allows only two identification attempts on one call (N=2). On the first failure, the number of tries is less than N, therefore processing returns to step S10, and the IP 23 again transmits the prompt for speech input. The caller again speaks the requested input information (S11), and the authentication module 233 again analyzes the input information (S12). If the second input adequately matches a stored subscriber's information in step S13, the processing flows through steps S14 to S18 to complete the call as described above.

However, if the extracted speech information does not match a stored subscriber template or feature data, processing again flows to step S19. If the number of tries now corresponds to the limit N, for example on the second failed attempt, the processing branches to step S20. The IP 23 may now transmit a message indicating denial of service, although this is optional. If provided, the message states that only a limited class of service is available in view of the problems in recognizing the caller as a known subscriber.

The IP 23 formulates a D-channel signaling message instructing the central office switch 11_1 to process the call in accord with default conditions and transmits that instruction to the central office switch (step S21). The instruction could include a default OE number corresponding to a default profile, or the message could instruct the switch to proceed using the OE and profile data for the off-hook line itself. The IP 23 supplies the message to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link (step S21) The administrative module processor 61 resumes call processing using the appropriate default OE and profile.

In the preferred embodiment, the switch provides a normal dial tone (S22), collects dialed digits from the caller (S23) and processes the call (S24). However, the default profile provides only some limited class of service, for example only emergency 911 service or 911 service plus flat rate local calling. The default call processing provides no additional information from the profile corresponding to any particular subscriber, therefore the network processes the call as a normal call for caller ID purposes. The caller ID service will provide only the telephone number to callers having normal caller ID, and the network will access LIDB database 21 to provide name information if any associated strictly with the telephone number, essentially in the manner that the network provides such services when there is no personalized dial tone service involved.

In the above example, the network disconnected the IP 23 after identifying the subscriber and providing the subscriber's virtual OE number to the serving central office 11. For some applications of the personal dial tone service, the central office 11 would maintain a bridged connection of the IP 23 on the line, to enable the IP to monitor the call. For example, in a prisoner telephone service, each prisoner would have only limited telephone rights as specified in each prisoner's profile data. To prevent one prisoner from selling their telephone service rights to another prisoner, the IP 23 would periodically or constantly monitor the outgoing speech signals from the prison line. The voice authentication module 233 would initially identify the prisoner subscriber as discussed above, and would periodically recheck to authenticate the identity of the party using the prison line. If the voice authentication module detects some other party using the line or did not detect the identified subscriber's speech for some predefined time interval, the IP 23 would instruct the serving central office switch 11 to disconnect the call. The IP 23 may send messages to the switch or to other network elements to initiate additional action, such as profile modification to further limit a particular prisoner's telephone privileges and/or to notify prison authorities of misuse of telephone privileges.

The first detailed example discussed above related to personal dial tone service provided on a per-call basis on a shared use line. Several known subscribers might routinely use their personal dial tone service over the same line. As noted earlier, an alternate form of the personal dial tone service can be activated on a dial-up basis. Consider now an example of a dial-up activation for a single call.

For this example, assume that a subscriber's normal or 'home' telephone is telephone 1_B. The end office switch 11_N stores the subscriber profile data for the line associated with that telephone station. Now assume that the subscriber is using station 1_A connected through a telephone line to central office 11_1. FIG. 5 provides a simplified flow diagram of the signal flow and processing for such a call.

The subscriber lifts the handset creating an off-hook state in the telephone 1_A and a signal to office 11 (step S31). The serving central office 11_1 detects the off-hook and commences its call processing. Specifically, the central office assigns a register in the call store 67 to this call and loads profile information associated with the off-hook line from the disc storage 63 into the register. In this case, the profile data associated with the line does not provide an off-hook trigger because the line is not specifically associated with the shared use type personal dial tone service discussed above. The central office 11_1 therefore provides dial tone in the normal manner (step S32).

If making a normal call, the caller would dial a destination number, and the network would complete the call as dialed. To activate the personal dial tone service, however, the subscriber dials an access number assigned to that service, such as 1-800-DIALTON, from the station 1_A (step S33).

The dialing of an outgoing call, in this case to the access number, is another type of service request. The central office switch 11_1 recognizes the dialed access number as a trigger event or 'PIC' (step S34). The SSP type central office 11_1 creates a TCAP query message containing relevant information, such as the office equipment (OE) number and/or telephone number assigned to the off-hook line, the dialed number and the type of triggering event. The office 11_1 transmits that query to the SCP 19 (step S35). Specifically, the SSP central office 11_1 transmits the query over an SS7 link to one of the STPs 15. The query includes a point code and/or a global title translation addressing the message to the SCP 19, and the STP 15 relays the query message over the appropriate link to the SCP 19.

In response to a query, the SCP 19 accesses its database to determine how to process the particular call. In this case, the query indicates the dialed number type trigger and provides the digits of the specific number dialed. The SCP 19 uses the dialed number as the access key. The SCP 19 retrieves a call processing record (CPR) corresponding to that number associated with the personal dial tone access function (step S36). For the current exemplary access, the CPR will provide information necessary for routing the call to the IP 23 that will perform the necessary speaker identification/verification (SIV).

Based on the CPR, the SCP 19 formulates a response message instructing the SSP central office 11_1 serving the customer to route the call. In this case, the message includes information, e.g. a office equipment (OE) number or telephone number, used for routing a call to the identified IP 23.

The SCP 19 formulates a TCAP response in SS7 format and transmits the TCAP response message back to the SSP central office 11_1 (see step S37).

The SSP type switch in the central office 11_1 uses the routing information to connect the call to a line or channel to the IP 23. A voice grade call connection now extends between the calling station 1_A and the IP 23 (step S38).

The central office 11 provides a signaling message to the IP 23 with the call. In this case, the signaling message includes the dialed digits indicating a call to the personal dial tone access number. The signaling message also includes either the office equipment number or the telephone number of the line to the calling station 1_A.

As in the earlier example, the IP 23 will seize the line for the incoming call and launch a query to the SCP 19 through the TCP/IP network 27 (step S39). The SCP 19 accesses an appropriate CPR (S40), and based on that CPR, the SCP 19 transmits back a message (S41) instructing the IP 23 to execute a program or script for the dial-up access to the personal dial-tone service.

The IP initially plays a greeting and a prompt message (S42) and collects spoken input information (S43). The IP 23 may also play a prompt and collect digits representing the subscriber's normal or home telephone number. The voice authentication module 233 analyzes the spoken identification information to extract characteristic information (S44) and compares the extracted information to stored template or feature data to determine if there is an adequate match to the known subscriber data (S45), as in the earlier example.

In step S45, the IP 23 determines if the information extracted from the speech input matches any of the stored template data feature data for an identifiable subscriber. If there is a match, the IP now knows the identity of the calling subscriber. Based on the identity of the subscriber, the IP 23 obtains the subscriber's profile data from the central office 11_W serving the subscriber's home telephone line. If the IP 23 is in direct signaling communication with the home central office 11_W, for example via an ISDN D-channel or an SMDI link, the IP 23 may directly request and receive the profile data over the signaling link. If the IP and the switch are not in direct communication, the IP may provide a message notifying the SCP 19, and the SCP 19 would obtain the data from the switch and provide it back to the IP 23.

The IP 23 formulates a D-channel signaling message containing the subscriber's profile information together with an instruction to load that information into the register assigned to the call in place of the profile information corresponding to the off-hook line (step S46). The IP 23 supplies the message to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link. In response, the administrative module processor 61 rewrites the profile data in the register assigned to the call using the data from the IP 23 (step S47). As such, the profile information in the assigned register now corresponds to the identified subscriber.

When the central office switch 11_1 reloads the profile, the central office disconnects the link to the IP 23 and connects tone receivers to the caller's line. The central office 11_1 may also provide a standard dial tone or other message over the line (step S48). The caller can now dial digits in the normal manner (step S49), and the switch in the central office 11_1 will load the dialed digits into the assigned register within the call store 67. The central office 11_1 utilizes the dialed digits and the subscriber's profile data to process the call (step S50). For example, the switch in central office 11_1 may provide the appropriate record to bill the outgoing call to the subscriber's account. In accord with the invention, the network also provides the subscriber specific information for caller ID purposes, in the manner discussed in detail above relative to either FIG. 4B or FIG. 4C.

As in the earlier example, the preferred embodiment allows up to N tries or attempts to provide recognizable subscriber identification information. Thus, if in step S45 the extracted information characterizing the input speech signals did not match any of the templates or feature data used by the IP 23, then the process flows to step S51. If the current number of attempts for recognition on this call is less than N, processing returns to step S42, and the IP 23 again transmits the prompt for speech input. The caller again speaks the requested input information (S43), and the authentication module 233 again analyzes the input information (S44). If the second input adequately matches a stored subscriber's information S45, the processing flows through steps S46 to S50 to complete the call as described above.

However, if the extracted speech information does not match a stored subscriber template or feature data, processing again flows to step S51. If the number of tries now corresponds to the limit N, the processing branches to step S52. The IP 23 preferably transmits a message indicating denial of service (S52), and then transmits a message to the central office 11_1 signifying disconnection of the access call (S53). It should be noted that, in this example, normal service provided over the line to station 1_A is available on a subsequent call. The failure to recognize the caller as a personal dial tone subscriber only prevents the caller from using the personal dial tone services of a subscriber to that service, for example specialized billing of calls to that subscriber's account instead of to the account normally associated with the line to the calling station 1_A.

In the above discussed dial-up access example, the dial tone service was personalized for a single outgoing call by temporarily loading the subscriber's profile data into the register assigned to the outgoing call in the originating central office 11_1. The system can provide such service to the subscriber over any line or to any telephone station, including pay telephone stations.

The present invention also enables activation of the personal dial tone service on a particular line for some predetermined period of time, for example to enable use of office or business services from some remote location while a business subscriber is out of the office. This type of operation involves an activation call requesting the service on a particular line for the desired period. Consider now an example of such a time activated service.

For this example, assume that a subscriber's normal business telephone is telephone 1_B. The end office switch 11_W stores the subscriber profile data for the line associated with that telephone station. Now assume that the subscriber is using station 1_A connected through a telephone line to central office 11_1 for business related communication services. The business related communication services include both incoming call related services and outgoing call related services.

To activate the personal dial tone service, the subscriber again lifts the handset at station 1_A, receives dial tone from the central office 11, and dials the access number assigned to that service. The network uses AIN type processing to route the call to the IP 23, as in the example discussed above relative to FIG. 5.

As in the earlier examples, the IP 23 seizes the line for the incoming call and launches a query to the SCP 19 through

the TCP/IP network 27. The SCP 19 transmits back a message instructing the IP 23 to play a greeting and a prompt message and collect and analyze spoken input information to identify and authenticate the subscriber. The instruction from the SCP 19 also causes the IP 23 to prompt the subscriber and obtain input information regarding the time period for service activation and possibly to obtain digits representing the subscriber's normal business telephone number. The process of calling the access number and interacting with the IP to activate the personal dial tone service on a line for the desired period is another type of service request.

For outgoing call processing, the IP 23 signals the central office 11_1 serving the line to station 1_A to set an off-hook trigger in the profile data associated with that line. The IP also obtains the profile information from the switch 11_N serving the station 1_B and provides that information together with a virtual OE number to the central office 11_1. The office 11_1 stores the profile in its disc memory 63 in such a manner that the switch in that office can use the virtual OE number to retrieve that subscriber's profile. For incoming calls to the subscriber, the IP 23 transmits a signaling message to the subscriber's home office 11_N to set a terminating attempt trigger (TAT) against the line to the subscriber's office telephone 1_B.

The IP 23 also transmits a message through the TCP/IP network 27 to the SCP 19 advising the SCP 19 of the service activation. This message identifies the subscriber, for example by their normal telephone number and identifies the telephone number and office equipment (OE) number associated with the line to station 1_A that the subscriber selected for their personal dial tone service.

In response to the message from the IP 23, the SCP 19 now establishes or modifies two CPRs for this subscriber. One CPR controls processing of calls to the subscriber's normal business telephone number to enable routing to the station 1_A, and the other controls routing of outgoing calls from that station to the IP 23 for speaker identification/verification (SIV) processing.

Subsequently, when there is an outgoing call from the station 1_A, the network will route the call to the IP 23 to determine if the caller is the subscriber or some other party, exactly as discussed in the per-call service from a shared use line (FIG. 4). As in that earlier example, if the IP identifies the caller as the personal dial tone subscriber, then the IP 23 provides the virtual OE number to enable loading of subscriber's profile from disc memory 63. The network provides the telephone number and the subscriber specific information, for caller ID purposes, as discussed above. If the IP determines that the caller is not the personal dial tone subscriber, the IP instructs the originating office 11_1 to simply provide dial tone and complete the call in the normal manner. The central office 11_1 therefore will utilize the office equipment (OE) number and profile information normally associated with the line, instead of those for the personal dial tone subscriber. The network provides caller ID service, identifying the number and possibly the main name associated with the line, in the normal manner. In this way, it is quite easy for the personal dial tone subscriber and the normal subscriber to both obtain their desired services on their respective calls via the same line, and to be correctly identified to called parties who subscribe to caller ID services.

The trigger set against the subscriber's normal telephone number and establishment of the CPR in the SCP 19 enables redirection of calls normally intended for the subscriber's business telephone 1_B to the line to station 1_A. Depending on how the subscriber elects to define their individual service, the network may simply route the calls to the line to station 1_A, as a normal AIN forwarded call that simply rings the station(s) 1_A on the line. Alternatively, the subscriber may elect an enhanced service which involves routing to the IP, IP prompting and speech recognition to identify the called subscriber and distinctive ringing over the line, in a manner analogous to that used for processing incoming calls in shared use applications, such as the above discussed dormitory example.

As noted above, the dial-up access procedure in this latest service example required the subscriber to specify a time period that the personal dial tone service should apply to the particular line. The IP 23 stores a record of the time period elected by the subscriber. When the period expires or if the subscriber calls in earlier to change the service to another line or temporarily cancel the service, the IP 23 will provide cancellation notices to the appropriate central offices 11 and to the SCP 19. In the example, the IP 23 will notify the office 11_1 to cancel the off-hook trigger set against the line to station 1_A and to delete the subscriber's virtual OE number and profile from its disc memory. The IP 23 will also instruct the central office 11_N to cancel the terminating attempt trigger set against the subscriber's business line to station 1_B. The notice to the SCP 19 causes the SCP to deactivate the personal dial tone CPR and the call redirection CPR. If the associated personal identification functionality for caller ID service relies on a central database, such as LIDB, the IP would also instruct that database to temporarily establish a subscriber account record associating the subscriber's name and calling card billing information with the telephone number and a subscriber identifier code.

The subscriber can then or later interact with the IP 23 to establish time based temporary personal dial tone service through another line or location, as discussed above. In this manner, a subscriber might set up a temporary office in a motel in one city for several days. The subscriber might cancel the service while in transit to a new location. Then the subscriber might reestablish the service to set up a temporary office service at a vacation home for a week.

The time based personal dial tone service could be modified in several manners. For example, the subscriber might establish a file for use by the SCP or the IP to establish the personal dial tone service at two or more locations at specified times, e.g. at the office during office hours and at a home office during other hours. Also, the above example of this service relied on downloading the subscriber' profile into the switch serving the line with which the subscriber is temporarily associated. Alternatively, the IP could obtain the profile from the subscriber's home switch and provide the profile to the serving switch as part of the processing of each outgoing call by the subscriber from that line during the specified time period.

A preferred network implementation and a number of specific call processing routines have been discussed above by way of examples relating to the present invention. However, the preferred embodiment of the invention is amenable to a variety of modifications.

For example, the preferred embodiment described above utilizes speaker identification/verification to recognize the identity of a calling subscriber. Where such capabilities are not available, the system could use an announcement and digit collection process, for example to obtain an account number and a personal identification number (PIN).

Also, the currently preferred embodiment utilizes AIN routing to the IP and speaker identification/verification ele-

ments within the IP to identify the subscriber for profile selection. As speaker identification/verification equipment becomes more readily available, cheaper and more compact, it will be possible to build this functionality into the line cards of the end office switches. The switch itself will challenge the caller, analyze spoken information and identify the subscriber to select the appropriate profile, without routing to an IP or the like.

While the foregoing embodiments of the invention supply many outstanding needs, there still exists a need for a method of conveniently and economically coping with a number of problems which manifest themselves in one or another objectionable type of usage of the public telephone network. These may comprise usages which are either illegal or detrimental to the health, safety and security of Telco subscribers. By way of example, one problem of widespread significance is the provision of adequate protection of the security and well being of so called "latchkey children." As will be understood, this term is applied to children, usually of school age, who have working parents but who arrive home from school prior to the return of their parents.

These children admit themselves to their residence or premises and are usually instructed by their parents to keep the door locked or latched until a parent returns. In addition to these instructions parents usually admonish such children to follow parent prescribed rules in answering or using the telephone. However, experience has demonstrated that the telephone is still subject to usages which pose threats of one or another types to the children. The problem is most acute where multiple children are housed with a single telephone link to the customer premises.

Parents or guardians usually provide each child with a list of permitted calls. For example, any of the children may be permitted to call 911 in case of emergency. All of the children may be permitted to call designated relatives or friends of the family. However, the call permissions and restrictions usually vary from child to child. The older children may be allowed calls to designated schoolmates or friends. The identity of the parties to whom the children are permitted to place calls varies with the identity of the child. Conversely each child may have individually prohibited calls. In the usual situation all calls which are not expressly designated as allowed will be prohibited.

In addition to this list of permitted outgoing calls, the children are usually provided with specific instructions as to calls to be answered. However this is difficult to regulate, even in the case of obedient children. For example, the availability of a Caller ID service offers no guarantee that the indicated caller is actually on the line. The present preferred embodiment of the invention provides a system and method for supplying this need.

Following is a description to the operation of one preferred embodiment of the invention which addresses the problem of providing implementation of the instructions of the parents or guardians with a reasonable degree of certitude.

In this example it is assumed that there is a subscriber premise which houses a pair of latchkey children A and B. Child A and child B have each been provided with a list of one or more incoming calls which they are permitted to receive. Each such child (hereinafter sometimes referred to as a subscriber) is provided with a personal customer profile record which is identified by a virtual OE number. Each such profile contains data which specifies permissible and prohibited communications for the individual child. For example, each customer profile identifies the callers whose calls may be accepted.

The central office switch identifies the particular line, by the OE number assigned to that line and line number. The switch also stores and retrieves profile data which it stores for that line and number and that profile data reflects the special services to which that line and number is subscribed. When the central office detects a call to a line having the personalized service, processing hits a terminating attempt trigger (TAT). The SSP switch interacts with the SCP and routes the call to the IP. The IP prompts the caller to identify a desired called party, e.g. one of the children sharing the line. Menu announcement together with either digit collection or preferably speech recognition processing by the IP facilitates identification of the desired called child from others associated with the line. Based on identification of the called child, the IP signals the SSP switch to load profile data for that specific child into the register assigned to the call in the call store.

This substitution is accomplished. In this case, the profile for child A contains data information which indicates that child A is permitted to accept a call from child C but that child C is required to authenticate herself. The IP is apprised of this requirement and uses another prompt to the calling party to identify herself. This may be a prompt such as "Who is calling?". A template for the voice of child C is maintained in the IP. This template is now used by the IP to verify that the caller is in fact child C. Child C has now been identified and authenticated as the calling party.

The profile for child A may provide that a distinctive ringing signal is to be used corresponding to the identified subscriber or child A. In this event distinctive ringing for child A is used to attempt to have child A answer the telephone. According to the loaded profile for child A, the answering party is prompted to speak her name. The IP remains bridged onto the connection and uses voice processing to verify a match between the spoken response and a template previously installed in the IP. Assuming verification, the switch concludes processing of the call in accord with the loaded profile information, i.e., makes the connection and permits the voice communication to occur. The IP is disconnected.

If the initial authentication of the calling party fails, i.e., if the caller states her name to be that of child C but the voice verification fails to confirm a match, the calling party may be permitted one or more additional attempts. If these fail, the invention comprehends a plurality of consequential handling steps.

In the simplest case the call attempt is terminated by disconnection of the calling line or link. As another option, the incoming call may be forwarded to a third party line, such as a pre-designated line to a parent or guardian. In instances satisfying applicable provisions of law, the parent or guardian may record the ensuing dialogue. The specific handling which is performed is contained in the data information in the profile which has been personalized for the subscriber represented by child A.

The foregoing example has dealt with affording protection to latchkey children in the case of incoming calls. It is a further feature of the invention that the invention provides a system for preventing the initiation of proscribed outgoing calls from the subscriber premise and line. Following is an example of the operation of such protection.

As previously stated, the preferred routing of the calls in accord with the invention utilizes AIN type call processing. In the case of one of the children A or B initiating a call the phone goes off-hook. The serving central office 11_1 detects the off-hook and commences its call processing.

Specifically, the central office assigns a register in the call store 67 to this call and loads profile information associated with the off-hook line from the disc storage 63 into the assigned register. In this case, the central office 11_1 is an SSP capable office, and the loaded profile data indicates an off-hook immediate trigger set against the particular line. The serving SSP type office 11_1 therefore detects this off-hook PIC as an AIN trigger.

In response to the off-hook and the off-hook trigger set in the subscriber's profile, the SSP type central office switch 11_1 launches a query to the SCP 19. Specifically, the SSP 11_1 creates a TCAP query message containing relevant information, such as the office equipment (OE) number assigned to the off-hook line, and transmits that query over an SS7 link to one of the STPs 15.

The STP 15 relays the query message over the appropriate link to the SCP 19. The query from the SSP central office 11_1 identifies the caller's line by its associated office equipment (OE) number and possibly by a single telephone number associated with the off-hook line.

In response to a query, the SCP 19 accesses its a database, typically, the MSAP database set up in the ISCP, to determine how to process the particular call. The SCP 19 identifies an access key in the query and uses the key to retrieve the appropriate record from the database. In this case, the query indicates an off-hook trigger as the trigger event, therefore the SCP 19 uses the calling party office equipment (OE) number as the access key. The SCP 19 retrieves a call processing record (CPR) corresponding to the office equipment (OE) number associated with the off-hook line and proceeds in accord with that CPR.

The CPR will provide information necessary for routing the call to some node of the network that will perform speaker identification/verification (SIV), in this example the SIV is a function performed by an Intelligent Peripheral (IP). Therefore the CPR provides information for routing the call to the nearest available IP having the SIV capability.

Based on the CPR, the SCP 19 formulates a response message instructing the SSP central office 11_1 serving the customer to route the call. In this case, the message includes information, e.g. a office equipment (OE) number or telephone number, used for routing a call to the identified IP 23. The SCP 19 formulates a TCAP message in SS7 format, with the destination point code identifying the SSP office 11_1. The SCP 19 transmits the TCAP response message back over the SS7 link to the STP 15, and the STP 15 in turn routes the TCAP message to the SSP central office 11_1.

The SSP type switch in the central office 11_1 uses the routing information to connect the call to one of the lines or channels to the IP 23. A two-way voice grade call connection now extends between the calling station 1_A and the IP 23. In the present example, the switch actually connects the off-hook line to the line to the IP before providing dial tone.

As noted above, the communication link to the IP 23 provides both line connections and signaling, preferably over a primary rate interface (PRI) type ISDN link. When the central office 11_1 extends the call from the calling party's line to a line circuit (over a B channel) to the IP 23, the switch in that office also provides call related data over the signaling link (D channel for ISDN). The call related data, for example, includes the office equipment (OE) number normally associated with the off-hook line and possibly the telephone number for that line.

In response to the incoming call, the IP 23 will seize the line, and it will launch its own query to the SCP 19 (step S7). In the preferred network illustrated in FIG. 1, the IP 23 and the SCP 19 communicate with each other via a separate second signaling network 27, for example utilizing either an 1129+ protocol or a generic data interface (GDI) protocol. The query from the IP 23 again identifies the caller's line by at least its associated office equipment (OE) number.

In response to the query from the IP 23, the SCP 19 again accesses the appropriate CPR and provides a responsive instruction back through the network 27 to the IP 23. Although the IP 23 could passively monitor any speech that the user might utter, the preferred implementation utilizes a 'Challenge Phase' to prompt the user to input specific identifying information. In this case, the instruction causes the IP 23 to provide a prompt message over the connection to the caller. Here, the signal to the caller is preferably an audio announcement prompting the caller to speak personal information. In one preferred example, the IP plays an audio prompt message asking the caller, 'Please say your name'. The process may ask for any appropriate identifying information.

The signal received by the IP 23 goes over the lines and through the central office switch(es) for presentation via the off-hook telephone 1_A to the calling party. In response, the caller will speak identifying information into their off-hook telephone, and the network will transport the audio signal to the IP 23. When the IP 23 receives speech input information during actual call processing, for this service example, the IP analyzes the speech to extract certain characteristic information.

As previously explained, the IP 23 stores a template or other voice pattern information for each person who has the personalized service in the area that the IP normally services. If the IP 23 does not store the particular template or feature information it needs to process a call, the IP 23 can communicate with a remote IP 23_R to obtain that information. In the present shared line example, the IP 23 will store template or feature data for each subscriber associated with the particular off-hook line.

When the IP 23 receives input speech and extracts the characteristic information during actual call processing, the IP compares the extracted speech information to stored pattern information, to identify and authenticate the particular caller. In the present example, the voice authentication module 233 in the IP 23 compares the extracted speech information to the stored template or feature data for each subscriber associated with the particular off-hook line. This includes the children A and B.

The IP 23 determines if the information extracted from the speech input matches any of the stored template data feature data for an identifiable subscriber. If there is a match, the IP now knows the identity of the calling subscriber. Based on the identification of the calling subscriber, the IP 23 selects a virtual office equipment (OE) number from storage that corresponds to the subscriber.

The IP 23 formulates a D-channel signaling message containing the virtual office equipment (OE) number together with an instruction to load that OE number into the register assigned to the call in place of the OE number of the off-hook line. The IP 23 supplies the message to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link. In response, the administrative module processor 61 rewrites the OE number in the register assigned to the call using the OE number received from the IP 23.

Upon rewriting the OE number in the register, the administrative module processor 61 of central office switch 11_1 also reloads the profile information in the register. Specifically, the administrative module processor 61

retrieves profile information associated with the virtual office equipment (OE) number from the disc storage 63 into the register. As such, the profile information in the assigned register in the call store 67 now corresponds to the identified subscriber, rather than to the off-hook line.

The profile information provides a wide range of data relating to the subscriber's services, including the permissions and restrictions applicable to the involved children. In the presently preferred implementation, when the central office switch 11_1 reloads the profile, the central office disconnects the link to the IP 23 and connects tone receivers to the caller's line. Optionally, the central office 11_1 may provide a 'dial tone' or other message over the line. The caller now dials digits in the normal manner, and the switch in the central office 11_1 loads the dialed digits into the assigned register within the call store 67. The central office 11 utilizes the dialed digits and the subscriber's profile data to process the call. If the dialed digits represent a call permitted to the caller further processing proceeds. On the other hand, if the number is not included in those which are permitted to the particular caller one of several alternative steps may follow. In the simplest situation the call processing may be discontinued with or without an audio announcement to the caller. As an alternative the call may be completed to a directory number supplied by the parent or guardian who then admonishes the child.

Assuming that the dialed digits match digits stored in the callers profile, it is a feature of the invention that actual verification of the authenticity of the responding party is performed. To this end an IAM message is sent to the destination SSP containing data in addition to that which is typically carried. This data information instructs the SSP to execute a pre-designated verification procedure. According to one preferred procedure the destination SSP sets up a voice connection between the IP and the called terminal. This is established via data signaling similar to that described in establishing a voice link between the originating central office and the IP for the originating end voice processing.

The availability of the called terminal is established by standard CCIS signaling, ringing signals are sent, and a responding party goes off-hook. Again a challenge prompt is delivered requesting the name of the responding party. When this is provided the signal is processed in the IP against a pre-prepared template which is mandated by the personal profile of the caller. Assuming a match is established, this is signaled by the IP to the originating switch and a trunk connection is established between the calling and called terminals. If no match can be established after a pre-specified number of attempts the caller is advised and the call processing discontinued. An audio announcement to the calling party is preferably provided.

According to yet another feature of the invention a system is provided for protecting the subscriber against the calls of stalkers or other recurring threatening calls. In this situation it is assumed that the unwanted calls have been received a sufficient number of times to allow the called line to record and create voice templates for the threatening caller. These may include the name or pseudonym used by the caller and optionally the name of the called party, where the stalker is calling for a particular person.

In the handling of this type of call pursuant to one preferred embodiment of the invention, the protected or guarded line and directory number have a terminating attempt trigger (TAT) set against the particular line. When the central office SSP which serves that line or local loop

detects a call (receives the IAM) to that line, it loads profile information associated with the called line. The loaded profile data indicates a terminating attempt trigger (TAT) set against the particular line. The SSP interacts with the SCP, and finds that identification of the called party is necessary. In accord with directions from the SCP the call is routed to the IP.

The IP prompts the caller to identify a desired called party. The IP uses speech recognition processing to identify the desired called party from those associated with the premises line. Based on identification of the called subscriber, the IP signals the SSP switch to substitute the CPR or profile of the now identified called party for the presently loaded CPR designated by the line OE. Thus the virtual OE profile of the specific called party is substituted for the line profile.

This substitution having been accomplished, the installed profile contains data information which indicates that the identity of the calling party is to be sought. The IP is apprised of this requirement and uses another prompt to the calling party to identify himself. This may be a prompt such as "Who is calling?". A template for the name of the harassing party is preferably maintained in the IP. This template is now used by the IP to verify that the caller is in fact the harasser or stalker. Alternatively the IP speech recognition module is trained to recognize the name.

The virtual OE profile of the called party contains data information for further handling of the call. A number of alternatives may be provided either singly or in combination. The profile may direct that the serving central office forward the call to a specified directory number of a third station. This station may constitute a terminal of a police authority or investigative organization. Police pursuit of the caller may ensue if sufficient information is available. The terminal may also record the ensuing dialog. In the case where a stalker situation pertains to minors, the call may be forwarded to a station of a parent or guardian. This option may occur either as an alternative or in addition to forwarding the call to the station of a police authority. Again recording may be performed if appropriate.

While the foregoing handling of an undesired call has been described in terms of a stalker, it will be understood that the methodology may be applicable to other types of calls. As a further example, pornographic calls may be intercepted and handled in a like manner. Assuming the satisfaction of applicable legal requirements evidence may be gathered to assist in criminal or civil legal proceedings.

In addition to the foregoing, the invention also comprehends providing assistance to authorities and police in the apprehension of wanted individuals. To this end the instant embodiment of the invention includes identification of the site of origination of the offending call. This information may then be automatically brought to the attention of the cognizant law enforcement authority on a real time basis. As a further alternative the information may be used to trigger other legally sanctioned investigatory or police action.

In the type of two party situation now under consideration the calling target individual may be subject to a court order prohibiting contact with the called party. In addition to the prohibition, the court order may authorize monitoring and recording of any prohibited calls and apprehension of the offending party. Such situations may be encountered with stalkers, pornographic callers, in troubled marital situations, among others. In the cases of stalkers and pornographic callers there are two party situations involving illegal activity, although the caller may be identified only by voice. However, monitoring and recording may be authorized as well as apprehension of the offender.

In the known stalker situation there may be an outstanding order for arrest which is not limited to apprehending the stalker when engaged in the illegal activity. However, the stalker may be known only by voice identification and the whereabouts of the stalker is usually unknown. Under such circumstances the methodology of this embodiment of the invention provides an opportunity to track the location of the stalker and to possibly gain further information to permit more specific identification. Identification of the site of the stalker or other such caller during actual engagement in a prohibited telephone conversation may offer the optimal possibility of effecting prompt apprehension. It will be appreciated that even though there may be an outstanding order for the arrest of the person involved, it is impractical, as well as violative of legal rights, to indiscriminately monitor a large number of telephone lines of uninvolved and uninformed parties. The two party situation offers both a legal and practical application for the use of the features of the present embodiment of the invention.

In one illustrative application of this embodiment of the invention, the aggrieved individual and associated subscriber terminal are known. This information identifies the subscriber line and its office equipment or OE number or designation, the aggrieved individual, and his or her virtual OE number. It is assumed that the speech processing facility, preferably an intelligent peripheral or IP has been provided with appropriate speech templates to implement speech identification and/or authentication of an offending voice identifiable individual. The virtual OE designation of the aggrieved individual identifies a personal customer profile record (CPR) for the aggrieved party. This profile defines procedures to be followed upon the aggrieved party receiving a call from a caller who has been previously identified by his or her speech as the wanted offending party.

An illustrative example of the operation of this embodiment of the invention under these circumstances is now described. In this case speech recognition is used with respect to the voice of the calling party. Current speech recognition technology permits recognition with a reasonable degree of certitude based on training from a limited sample of recorded speech of a subject. In a situation of this type the target of the speech recognition is a wanted person and is not likely to cooperatively participate in any type of prompting procedure which may seem suspicious. As a result, it may be necessary to rely on such sophisticated speech recognition techniques as applied to random speech. On the other hand, in some instances it may be found possible to obtain speech templates of the target person uttering the name of the aggrieved party. This alone may prove sufficient to provide the necessary identification.

In such situations the instant embodiment of the invention relies on the speech recognition capability of the module 235 in the IP 23. The speech recognition module 235 enables the IP to analyze incoming audio information to recognize vocabulary words. The IP interprets the spoken words and phrases to determine subsequent action. For example, the IP may recognize the target caller speaking the name of a called subscriber and use the subscriber identification obtained in this manner to instruct the terminating central office to thereafter control the call in accord with that named subscriber's profile. On the other hand, the target caller may simply go on hook if he or she does not recognize the voice of the aggrieved party answering the telephone, so that another speech recognition procedure is necessary.

In this example the offending or target party may go off hook at any telephone in the relevant network. This creates a corresponding signal or change in state on the line to the central office to which the telephone is connected. In the call sequence, the off-hook signal acts as a type of service request, i.e. a request to make an outgoing call. The originating central office detects the off-hook and commences its call processing. Specifically, the originating central office assigns a register to the call in the call store of the originating central office switch. The switch loads profile information associated with the off-hook line from the disc storage of the switch into the assigned register.

The originating central office provides dial tone or the like over the line, the caller dials digits corresponding to the desired destination and the switch in the originating central office begins its processing to route the call through the network. The originating central office uses the dialed number to initiate a CCIS communication with the exchange serving the intended destination, in this example the terminating central office, which is assumed to be an SSP central office.

Specifically, the originating central office generates an Initial Address Message (IAM) for transmission to the terminating central office. The IAM message includes the SS7 destination point code (DPC) of the terminating central office and the SS7 origination point code (OPC) of the originating central office for addressing purposes. The payload portion of the IAM message includes the called and calling numbers. The originating central office transmits the IAM message through the CCIS network to the distant terminating office.

When the terminating office receives the IAM message, the administrative module processor for that office retrieves the customer profile for the number in the destination number field of that message (e.g. the number for the telephone line identified by OE in the destination central office for that number), from its mass storage system and loads that profile into one of its call store registers.

The subscriber for that line has personal dial tone service and a virtual OE assignment for each individual associated with that service. Usually such individuals reside at the site or subscriber premises at which the line or local loop terminates. The loaded profile for the OE of the line itself indicates a terminating trigger for that line and OE. The office of the subscriber, being an SSP type office, detects this call PIC as an AIN trigger.

In response to the IAM and the terminating attempt trigger (TAT) set in the subscriber's profile, the SSP type terminating central office switch launches a query to the SCP. Specifically, the SSP creates a TCAP query message containing relevant information, such as the office equipment (OE) number assigned to the called number line, and transmits that query over an SS7 link to one of the STPs. The query includes a destination point code and/or a global title translation addressing the message to the SCP, and the STP relays the query message over the appropriate link to the SCP. The query from the SSP terminating central office identifies the called line by its associated office equipment (OE) number and possibly by a single telephone number associated with the called line.

In response to a query, the SCP accesses its a database, typically, the MSAP database set up in the ISCP, to determine how to process the particular call. The SCP identifies an access key in the query and uses the key to retrieve the appropriate record from the database. In this case, the query indicates a terminating attempt trigger as the trigger event, therefore the SCP uses the called line office equipment (OE) number as the access key. The SCP retrieves a call processing record (CPR) corresponding to the office equipment

(OE) number associated with the called line and proceeds in accord with that CPR.

The call processing by the destination central office switch utilizes the loaded subscriber profile information. In this instance the called station subscribes to personal dial tone service and the line OE is associated with multiple virtual OEs assigned to individuals residing at the same subscriber premises. Thus the subscriber profile of the OE specifies procedures to be followed to implement the particular personal dial tone service desired by the subscriber.

In the instant preferred embodiment of the invention a situation exists wherein an individual to whom one of the virtual OEs is assigned is being harassed or threatened by an individual who has been identified by voice only. The full identity and location of that target individual is not known. However, the threatened individual has been repeatedly harassed by telephone at the subscriber premises via the subscriber line. It is desired to locate and apprehend the offending party.

In order that this may be accomplished according to this preferred embodiment of the invention, it is desired to have the offender making a call to the threatened party speak a specific or sufficiently extended utterance to permit the calling party to be identified as the offending individual. According to one procedure, the CPR of the subscriber line OE for the premises may contain instructions to deliver a prompt to all callers. That prompt could request all such callers to speak the name of the called party. In this procedure the IP is equipped with speech templates to permit speech identification/verification or SIV of a specific virtual OE from the utterance of the name by the caller. However, if the caller is the offending party, he or she is likely to be wary and may be suspicious of such a prompt delivered request. In that event the party may simply go on-hook and thwart any possibility of identification or apprehension.

A more preferred procedure, pursuant to one feature of this embodiment of the invention, is to instruct all residents at the subscriber premises to always answer the telephone by speaking their own name, such as, "This is Jane" or "This is John," as the case may be. The IP has been provided with voice trained templates to enable identification of the desired virtual OE of each named individual from such a name utterance.

In response to the IAM, and to the terminating attempt trigger (TAT) set in the subscriber's profile, and to the procedures specified in the subscriber's profile, the SSP switch routes the call to the nearest IP having the necessary SIV capability. A two way voice grade call connection now extends between the called line and the IP. As noted above, the communication link to the IP 23 provides both line connections and signaling, preferably over a primary rate interface (PRI) type ISDN link. When the central office 11, extends the call from the calling party's line to a line circuit (over a B channel) to the IP 23, the switch in that office also provides call related data over the signaling link (D channel for ISDN). The destination central office switch now provides ringing signal to the called line via its OE. As a result one of the subscribers at the premises goes off-hook. It is optional as to whether or not the calling party is voice connected at this stage.

The next action is dependent on the identity of the person at the subscriber premises who answers the telephone. In this example it is assumed that the threatened or harassed individual answers. As previously stated, all answering parties are instructed to first identify themselves. Here the threatened answering party does so, as for example, "This is Jane." When the IP receives input speech and extracts the characteristic information during actual call processing, it compares the extracted speech information to stored pattern information to identify and authenticate the particular answering party or subscriber. In the present example, the voice authentication module 233 in the IP 23 compares the extracted speech information to the stored template or feature data for each subscriber associated with the particular off-hook line. The IP now knows the identity of the called subscriber. Based on the identification of the called subscriber, the IP 23 selects a virtual office equipment (OE) number from storage that corresponds to the subscriber.

The IP 23 formulates a D-channel signaling message containing the virtual office equipment (OE) number together with an instruction to load that OE number into the register assigned to the call in place of the OE number of the off-hook line. The IP 23 supplies the message to the SSP central office switch 11, over the D-channel of the ISDN PRI link (step S14). In response, the administrative module processor 61 rewrites the OE number in the register assigned to the call using the OE number received from the IP 23.

Upon rewriting the OE number in the register, the administrative module processor 61 of central office switch 11, also reloads the profile information in the register. Specifically, the administrative module processor 61 retrieves profile information associated with the virtual office equipment (OE) number from the disc storage 63 into the register. As such, the profile information in the assigned register in the call store 67 now corresponds to the identified subscriber, rather than to the off-hook line.

The profile information provides a wide range of data relating to the subscriber's services. Included in that information the profile data provides necessary instructions to alert the IP to prepare to attempt to match the speech of the calling party to identify the calling party as the target. If the procedure which is specified in the CPR includes the option of not voice connecting the calling party to the called line to hear the live response of the answering party subscriber, the IP is directed to record the response for playback to the calling party when that party has been voice connected. The calling party is now connected to the called line, which has a voice connection to the IP, i.e., it is bridged to the IP. The IP plays the recorded response to the calling party and prepares to monitor the speech of the caller and attempt to match it to that of the target.

If the IP is able to establish a match through its SIV procedures, the conversation is recorded. In addition, and as an example of procedures which may be specified according to one preferred embodiment of the invention, the loaded profile directs that the destination SSP interact with the IP to use the identity of the calling telephone number to identify the site of the calling telephone, and to send that information to the police. It is assumed that the police have been previously alerted to the situation and have authority and orders to apprehend the offending individual. As an attempt to provide the police with maximum time to respond, the threatened individual may have been requested to attempt to hold the offending party on the line as long as possible. The IP may be maintained in a bridged condition in order to monitor and record the conversation as a legal evidence procurement measure. In addition the conversation may also be bridged to the police.

The forwarding of the information locating the calling telephone may be implemented as follows. The loaded profile of the harassed party includes data that directs and initiates a sequence such as the following. The originating

end office has addressed and transmitted the IAM message with the called and calling telephone numbers. The terminating end office launches a second query message through one or more of the STP(s) to the LIDB database 21 in FIG. 1. The query message includes both the telephone number associated with the calling station or its telephone line as well as a code identifying the virtual OE and nature of this request.

The LIDB database uses the calling party telephone number and the code received in the query, to retrieve that calling subscriber terminal or line account file record from the database. This includes the name and address of the subscriber for that telephone station. The LIDB database 21 compiles a TCAP call control message including the name and address data and returns that call control message to the terminating central office via the SS7 network. The terminating central office switching system receives the call control message from the LIDB database and shares this information with the IP via the signaling circuit. The same information is transmitted to the police via signaling and/or voice circuit. The police are now bridged onto the voice conversation and have been provided with the address from which the offending call is being conducted. If a patrol car is in the vicinity of the identified address, or if the harassed person is able to hold the offending caller on the line a sufficient time, the car may be radio dispatched and an apprehension may be accomplished.

In the foregoing example it was assumed that the harassed party answered the telephone. The profile of that person was then installed based on that person identifying his or herself in answering the telephone. In the case where the telephone is answered by a resident of the premises other than the harassed party, that party answers with the same prespecified greeting, such as, "This is John." In this situation the profile in the CPR corresponding to the subscriber line may specify that the profile of that individual, namely John, be substituted for the line CPR. Unless the caller specifically requests to speak to the harassed party by name, the call will proceed in accord with the CPR of the answering party, in this instance, John.

If the caller, as yet unidentified, asks to speak to the harassed party, Jane in this example, alternative procedures may be utilized according to this embodiment of the invention. According to a first procedure, the answering party, John, calls the harassed party, Jane, in such a manner that his call for Jane is fully audible to the telephone microphone. This utterance is identified by the bridged IP SIV, and the CPR of Jane is substituted by the destination central office switch. Jane answers in the pre-agreed format, such as, "This is Jane." The CPR for Jane has been entered and the same procedure is followed as has been previously described in the instance in which Jane answered the telephone.

As an alternative to this procedure, an answering party other than Jane may say "Please hold," place the call on hold, and call Jane to the telephone. Jane may then remove the call from hold, and answer in the pre-agreed manner, such as, "This is Jane." The CPRs of all residents of the subscriber premises may contain call handling instruction data directing interpretation of hold signals on this line as directing connection to the SIV facilities of the IP and directing the IP to stand by to implement CPR selection corresponding to the next matching name of a subscriber. As a result the switch substitutes the CPR for Jane on identifying her name through SIV. The CPR for Jane is now entered and the same procedure is followed as has been previously described in the instance in which Jane answered the telephone. As a still further alternative to the foregoing, all subscriber OE profiles at the subscriber premises may contain processing instructions to cause the IP to be connected upon execution of a *HOLD sequence. In this case all subscribers are instructed to use a *HOLD sequence when calling the threatened party to the telephone.

Reference is now had to FIG. 6 to facilitate description of yet another embodiment of the inventions. The present day popularity of voice mail service has been previously mentioned in discussing the prior art background of the present inventions. There has also been mention of the Intelligent Peripheral or IP 23 (FIGS. 1 and 3) having a voice mail server 239 for use by the network illustrated in FIG. 1. FIG. 6 provides a diagrammatic depiction of one available voice mail server suited for use in a preferred embodiment of an implementation for providing voice mail services in the present multi-subscriber per line environment.

Referring to FIG. 6, the centralized message service or voice mail system in the illustrated example comprises voice messaging equipment such as a voice mail system 120. The voice mail system 120 includes a digital switching system (DSS) 121, a master control unit (MCU) 123, a number of voice processing units (VPUs) 125 and a master interface unit (MIU) or concentrator 127. The master control unit (MCU) 123 of the voice mail system 120 is a personal computer type device programmed to control overall operations of the system 120.

Each of the voice processing units 125 also is a personal computer type device. The voice processing units 125 each include or connect to one or more digital mass storage type memory units (not shown) in which the actual messages are stored. The mass storage units, for example, may comprise magnetic disc type memory devices. Although not specifically illustrated in the drawing, the voice processing units 125 also include appropriate circuitry to transmit and receive audio signals via T1 type digital audio lines. An ETHERNET type digital network 129 carries data signals between the MCU 123 and the voice processing units 125. The Ethernet network 129 also carries stored messages, in digital data form, between the various voice processing units 125. The system 120 further includes T1 type digitized audio links 128 between the DSS switch 121 and each of the voice processing units 125.

The voice mail system 120 connects to the central office switching system 11 via the network 240 and the direct talk modules 231A and 231B and ISDN PRI TRUNKS which provide voice and signaling channels. Communication with the SSPs in the central offices may also be had via the network 240, IP communication server 243 and router 241. The MIU 127 is a data concentrator which effectively provides a single connection of many data signal links into the MCU 123 of the voice mail system.

The above described voice mail system architecture is similar to existing voice mail type central messaging systems, such as disclosed in U.S. Pat. No. 5,029,199 to Jones et al., although other messaging system architectures such as disclosed in the other patents cited above could be used. See also U.S. Pat. No. 5,661,782 to Farris and Bartholomew for additional description of operation of this type of voice mail system.

For each party who subscribes to a voice mail service provided by the centralized messaging system 120, the MCU 123 stores information designating one of the voice processing units 125 as the "home" unit for that subscriber. Each voice processing unit 125 stores generic elements of prompt messages in a common area of its memory. Personalized elements of prompt messages, for example recorded

representations of each subscriber's name spoken in the subscriber's own voice, are stored in designated memory locations within the subscriber's "home" voice processing unit.

In voice mail systems of the type discussed above, a subscriber's "mailbox" does not actually correspond to a particular area of memory. Instead, the messages are stored in each "mailbox" by storing appropriate identification or tag data to identify the subscriber or subscriber's mailbox to which each message corresponds.

Each time a call comes in to the voice mail system 120, the master control unit 123 controls the digital switching system 121 to provide a multiplexed voice channel connection through to one of the voice processing units 125. Typically, the call connection goes to the "home" voice processing unit for the relevant subscriber. The voice mail subscriber is identified by data transmitted from the switching system 11, as described above, if the call is a forwarded call. If all 24 T1 channels to the "home" voice processing unit are engaged, the central processing unit 123 controls switch 121 to route the call to another voice processing unit 125 which is currently available.

The voice processing unit connected to the call retrieves prompt messages and/or previously stored messages from its memory and transmits them back to the calling party via the internal T1 line 128, the DSS switch 121 one of the voice channels, central office switching system 11 and the calling party's telephone line. The voice processing unit 125 connected to the call receives incoming messages from the caller through a similar route and stores those messages in digital form in its associated mass storage device.

When the incoming call is a forwarded call, the connected voice processing unit 125 provides an answering prompt message to the caller, typically including a personalized message recorded by the called subscriber. After the prompt, the voice processing unit 125 records a message from the caller and identifies that stored message as one for the called subscriber's mailbox.

At times the connected voice processing unit 125 will not have all necessary outgoing messages stored within its own associated memory. For example, a forwarded call normally will be connected to the called subscriber's "home" voice processing unit 125, but if the home unit is not available the forwarded call will be connected to a voice processing unit 125 other than the subscriber's home voice processing unit. In such a case, the connected unit 125 requests and receives from the home unit 125 the personalized components of the answering prompt message via the data network 129. The connected voice processing unit 125 will store the transferred message data in its own memory, and when necessary, will play back the transferred data from its own memory as outgoing messages in the exact same manner as for any prompts or greeting messages originally stored in its own memory.

The connected voice processing unit 125 also will store any incoming message in its own associated memory together with data identifying the message as one stored for the called subscriber's mailbox. As a result, the system 120 actually may store a number of messages for any given subscriber or mailbox in several different voice processing units 125. Subsequently, when the voice mail subscriber calls in to the voice mail system 120 to access the subscriber's mailbox, the call is connected to one voice processing unit 125. Again, this call typically goes to the home unit 125 but would go to a different available one of the units 125 if the home unit is not available at the time. In response to appropriate DTMF control signals, or preferably voice signals, received from the subscriber, the connected voice processing unit retrieves the subscriber's messages from its own memory and plays the messages back to the subscriber. If any messages are stored in other voice processing units, the connected unit 125 sends a request the other units 125 to download any messages for the subscriber's mailbox those units have actually stored. The downloaded messages are stored in the memory of the connected voice processing unit 125 which replays them to the subscriber.

In a typical usage of the present embodiment of the invention there may be four subscribers for the telephone station 1_A which is connected to a line or local loop having a single telephone number and office equipment or OE number. Each of the four subscribers is provided with a personalized service profile or customer profile (CPR) which is identified by a virtual office equipment or OE number. As previously explained a virtual office equipment number refers to "virtual" equipment which has no real existence in the relevant central office. The profiles include for each subscriber a range of information relating to subscribers services, such as service features, classes of service, individual billing options, information relating to restrictions applied to individual users, as well as the performance of functions related to that user. Each of the four subscribers in this example subscribes to voice mail service and is assigned a mail box or partition of a mail box in the voice mail server 239 in the intelligent peripheral or IP 23. It will be understood that while the voice mail unit or facility is here shown as integrated into the IP this need not be the case. The voice mail unit may be provided as a stand alone unit as shown, for example in the above cited Jones et al. Patent. The subscription to the voice mail service and identification of a mail box is included in the information in each customer profile relating to the specific subscriber. Similarly the information data may prescribe that a call is to be forwarded to the voice mail system on a 'busy' and/or 'no-answer' condition.

When the serving central office SSP 11 detects a call to a line having the personalized service, processing hits a terminating attempt trigger (TAT). The SSP interacts with the SCP 19 and routes the call to the IP 23. The IP 23 prompts the caller to identify a desired called party, e.g. one of the subscribers sharing the single line. Menu announcement together with speech or voice utterance recognition processing by the IP 23 enables identification of the desired called party from those subscribers associated with the line. Based on identification of the called subscriber, the IP 23 signals the SSP switch 11 to load profile data for that subscriber into the register assigned to the call in the call store. The switch 11 thereupon uses the selectively loaded personal profile information for terminating the call. The IP disconnects, and the SSP central office 11 processes the call in accord with the loaded profile information which is identified by the virtual OE of the called and now identified subscriber. As an alternative or conjunctively with this procedure, the SSP may first load into the assigned register a generic customer profile for the OE for the line itself. Such a service profile may contain information which is generic to the subscribers served by that line and OE umber, particularly subscribers who may not subscribe to any enhanced services.

For example, in accord with the now loaded profile the central office 11 may provide a distinctive ringing signal corresponding to the identified subscriber. This service enables distinctive ringing for multiple subscribers on one line without assigning each subscriber a separate telephone number. In this example the loaded profile information

specifies forwarding of the call to the identified subscriber's identified mailbox in event of a busy or no-answer condition.

It is a feature of this embodiment of the invention that the call is initially routed to the intelligent peripheral. The IP includes a so-called 'challenge' wherein the caller is requested to speak his or her name. The intelligent peripheral speech identification capability includes a previously obtained template or templates to permit identification of the called party through use of that template or equivalent speech identification/verification procedure. Upon making the identification the IP selects a virtual office equipment (OE) number from the storage that corresponds to the subscriber. The IP then sends instructions to load that OE number into the register assigned to the call in place of the OE number of the subscriber premises line or local loop. In response, the administrative module processor 61 rewrites the OE number in the register assigned to the call using the OE number received from the IP 23.

Upon rewriting the OE number in the register, the administrative module processor 61 of central office switch 11_1 also reloads the profile information in the register. Specifically, the administrative module processor 61 retrieves profile information associated with the virtual office equipment (OE) number from the disc storage 63 into the register. As such, the profile information in the assigned register in the call store 67 now corresponds to the identified subscriber, rather than to the customer premises line. Having loaded the proper customer profile record in the assigned call store register the destination central office utilizes the dialed digits and the now loaded subscriber's profile data to process the call. Thus, the generic profile corresponding to the OE number ('real OE number' for actual equipment) may be replaced by the profile corresponding to a virtual OE number.

Pursuant to the loaded profile the central office transmits to the called premises line a distinctive ringing signal which identifies the subscriber corresponding to the virtual OE number and profile. Also pursuant to the profile a no answer (or busy/no answer) condition results in forwarding the call to the intelligent peripheral via the voice and signaling links. When the call comes in to the voice mail system, the master control unit 123 controls the digital switching system 121 to provide a multiplexed voice channel connection through to one of the voice processing units 125. Typically, the call connection goes to the "home" voice processing unit for the relevant subscriber. The voice mail subscriber is identified by data transmitted from the switching system 11, as described above.

The voice processing unit connected to the call retrieves prompt messages and/or previously stored messages from its memory and transmits them back to the calling party via the internal T1 line 128, the DSS switch 121 one of the voice channels, central office switching system 11 and the calling party's telephone line. The voice processing unit 125 connected to the call receives incoming messages from the caller through a similar route and stores those messages in digital form in its associated mass storage device.

The connected voice processing unit 125 provides an answering prompt message to the caller, typically including a personalized message recorded by the called subscriber. After the prompt, the voice processing unit 125 records a message from the caller and identifies that stored message as one for the called subscriber's mailbox. Subsequently, when the voice mail subscriber calls in to the voice mail system 120 to access the subscriber's mailbox, the call is connected to one voice processing unit 125. Again, this call typically goes to the home unit 125 but would go to a different available one of the units 125 if the home unit is not available at the time. In response to appropriate DTMF control signals received from the subscriber, the connected voice processing unit retrieves the subscriber's messages from its own memory and plays the messages back to the subscriber. If any messages are stored in other voice processing units, the connected unit 125 sends a request the other units 125 to download any messages for the subscriber's mailbox those units have actually stored. The downloaded messages are stored in the memory of the connected voice processing unit 125 which replays them to the subscriber.

According to another feature of this embodiment of the invention the subscriber's profile contains data which prescribes a distinctive interrupted dial tone to indicate the presence of stored voice mail. Each of the four subscribers in this example subscribes to voice mail service and is assigned a mail box or partition of a mail box in the voice mail server 239 in the intelligent peripheral or IP 23. Each subscriber may also be assigned a distinctive interrupted dial tone, such as, for example, two short tones, a long and a short, etc., to indicate the presence of voice mail. However, such a distinctive tone is not necessary as will now become apparent.

As has been previously described, when one of the subscribers in the relevant multi-subscriber per line premises goes off-hook, this is interpreted as a request to make an outgoing call. The associated central office commences its call processing. Specifically, the central office assigns a register in the call store 67 to this call and loads profile information associated with the off-hook line from the disc storage 63 into the assigned register. In this case, the central office 11_1 is an SSP capable office, and the loaded profile data indicates an off-hook immediate trigger set against the particular line. The serving SSP type office 11_1 therefore detects this off-hook PIC as an AIN trigger.

In response to the off-hook and the off-hook trigger set in the subscriber's profile, the SSP type central office switch 11_1 launches a query to the SCP 19. Specifically, the SSP 11_1 creates a TCAP query message containing relevant information, such as the office equipment (OE) number assigned to the off-hook line, and transmits that query over an SS7 link to one of the STPs 15. The STP relays the query message over the appropriate link to the SCP 19. The query from the SSP central office 11_1 identifies the caller's line by its associated office equipment (OE) number and possibly by a single telephone number associated with the off-hook line.

In response to a query, the SCP 19 accesses its a database, typically, the MSAP database set up in the ISCP, to determine how to process the particular call. The SCP 19 identifies an access key in the query and uses the key to retrieve the appropriate record from the database. In this case, the query indicates an off-hook trigger as the trigger event, therefore the SCP 19 uses the calling party office equipment (OE) number as the access key. The SCP 19 retrieves a call processing record (CPR) corresponding to the office equipment (OE) number associated with the off-hook line and proceeds in accord with that CPR.

The CPR provides the information necessary for routing the call to a node of the network that will perform speaker identification/verification (SIV), in this example the Intelligent Peripheral or IP. Based on the CPR, the SCP 19 formulates a response message instructing the SSP central office 11_1 serving the customer to route the call. In this case, the message includes an office equipment (OE) number or

telephone number, used for routing a call to the identified IP 23. The SCP 19 formulates a TCAP message in SS7 format, with the destination point code identifying the SSP office 11_1. The SCP 19 transmits the TCAP response message back over the SS7 link to the STP 15, and the STP 15 in turn routes the TCAP message to the SSP central office 11_1.

The SSP type switch in the central office 11_1 uses the routing information to connect the call to one of the lines or channels to the IP 23. A two-way voice grade call connection now extends between the calling station 1_A and the IP 23. In the present example, the switch actually connects the off-hook line to the line to the IP before providing dial tone.

As noted above, the communication link to the IP 23 provides both line connections and signaling, preferably over a primary rate interface (PRI) type ISDN link. When the central office 11_1 extends the call from the calling party's line to a line circuit (over a B channel) to the IP 23, the switch in that office also provides call related data over the signaling link (D channel for ISDN). The call related data, for example, includes the office equipment (OE) number normally associated with the off-hook line and possibly the telephone number for that line.

In response to the incoming call, the IP 23 seizes the line, and launches its own query to the SCP 19. In the preferred network illustrated in FIG. 1, the IP 23 and the SCP 19 communicate with each other via the separate second signaling network 27. The query from the IP 23 again identifies the caller's line by at least its associated office equipment (OE) number.

In response to the query from the IP 23, the SCP 19 again accesses the appropriate CPR and provides a responsive instruction back through the network 27 to the IP 23. The IP issues a 'Challenge Phase' to prompt the user to input specific identifying information. In this case, the instruction causes the IP 23 to provide a prompt message over the connection to the caller. Here, the instruction from the SCP 19 causes the IP 23 to provide an audio announcement prompting the caller to speak personal information. In one preferred example, the IP plays an audio prompt message asking the caller, 'Please say your full name'. As previously explained the process may request any appropriate identifying information.

The signal received by the IP 23 goes over the lines and through the central office switch(es) for presentation via the off-hook telephone 1_A to the calling party. In response, the caller speaks identifying information into their off-hook telephone, and the network transports the audio signal to the IP 23.

As noted above, an IP 23 can provide a wide range of call processing functions. In this example, the IP performs speaker identification/verification (SIV) on the audio signal received from the off-hook telephone. When the IP 23 receives speech input information during actual call processing, for this service example, the IP analyzes the speech to extract certain characteristic information. As has been described above the IP stores templates or feature data for each subscriber associated with the particular off-hook line.

When the IP 23 receives input speech and extracts the characteristic information during actual call processing, the IP compares the extracted speech information to stored pattern information, to identity and authenticate the particular caller. In the present example, the voice authentication module 233 in the IP 23 compares the extracted speech information to the stored template or feature data for each subscriber associated with the particular off-hook line.

The IP 23 determines if the information extracted from the speech input matches any of the stored template data feature data for an identifiable subscriber. If there is a match, the IP now knows the identity of the subscriber who went off-hook. Based on the identification of the calling subscriber, the IP 23 selects a virtual office equipment (OE) number from storage that corresponds to the subscriber.

The IP 23 formulates a D-channel signaling message containing the virtual office equipment (OE) number together with an instruction to load that OE number into the register assigned to the call in place of the OE number of the off-hook line. The IP 23 supplies the message to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link. In response, the administrative module processor 61 rewrites the OE number in the register assigned to the call using the OE number received from the IP 23.

Upon rewriting the OE number in the register, the administrative module processor 61 of central office switch 11_1 also reloads the profile information in the register. Specifically, the administrative module processor 61 retrieves profile information associated with the virtual office equipment (OE) number from the disc storage 63 into the register. As such, the profile information in the assigned register in the call store 67 now corresponds to the identified subscriber, rather than to the off-hook line.

As described, the profile information provides a wide range of data relating to the subscriber's services. Included in this information is identification of this subscriber's mailbox and instructions for returning to the off-hook party a dial tone indicative of the presence of unread voice mail in that party's mailbox. This signal is delivered to the party who can then retrieve the message in the usual manner.

As a further feature of this embodiment of the invention, the subscriber profile may also contain instructions to permit a caller to elect to call directly to the mail box of any subscriber to the multi-subscriber single line' service. According to this feature the SIV facility, in this instance the IP, is provided with dual templates for each subscriber. As an example, one of the two templates may identify the subscriber's name, whereas the other template may identify the subscriber's name with the addition of a command such as "mailbox." When a caller speaks the full command "John Doe mailbox," the SIV facility interprets this as an identification and authorization to connect the call directly to the mailbox. In such a situation a ringing signal would be optional when the called line is free for use. When the line is busy the call would go directly to the IP and mailbox without the necessity for the prompt announcing the availability of a mailbox service.

While the foregoing has described what are considered to be preferred embodiments of the invention, it is understood that various modifications may be made therein and that the invention may be implemented in various forms and embodiments, and that it may be applied in numerous applications, only some of which have been described herein. It is intended by the following claims to claim all such modifications and variations which fall within the true scope of the invention.

What is claimed is:

1. A method comprising:

detecting a request to make a voice call from a first link to an identified second link through a communication network including multiple central office switching systems connected to multiple links;

identifying one of multiple parties available via said second link by processing signals resulting from speech transmitted via said first link;

selecting a customer profile record corresponding to a virtual office equipment designation assigned to said one party;

installing said customer profile record at a central office switching system serving said second link;

pursuant to information in said customer profile record storing a voice message in message storage associated with said second link and said one party for retrieval by said one party.

2. A method according to claim 1 including the step of transmitting said stored message responsive to identification by processing signals resulting from an utterance of a party retrieving said message.

3. A method according to claim 2 wherein said transmitting said stored message is responsive to said customer profile record following its reinstallation at said central office switching system serving said second link.

4. A method according to claim 3 wherein said identification by processing signals resulting from an utterance of a party retrieving said message is at least partially responsive to information contained in a customer profile record different than said customer profile record corresponding to a virtual office equipment designation assigned to said one party.

5. A method according to claim 4 wherein said reinstallation of said customer profile record is responsive to said identification by processing signals resulting from an utterance of a party retrieving said message.

6. A method according to claim 1 wherein said storing of a voice message is responsive to a busy or no answer condition at said second link.

7. A method according to claim 6 wherein said storing of a voice message is responsive to a voice prompt responsive to detection of said busy or no answer condition at said second link.

8. A method according to claim 1 wherein said storing of said voice message is responsive to processing of speech signals transmitted from said first link.

9. A method according to claim 8 wherein said last named speech comprises at least a portion of said speech transmitted via said first link identifying said one of multiple parties available via said second link.

10. A method according to claim 1 wherein each of said multiple parties is assigned a personalized customer profile record corresponding to an individual virtual office equipment designation.

11. A method comprising:

detecting a request to make a voice call from a first link to a second link through a switched telecommunication network including multiple central office switching systems connected to multiple links;

installing a first customer profile record corresponding to an office equipment designation assigned to said second link in a central office switching system serving said second link:

pursuant to information in said first customer profile record establishing a speech connection between said first link and a speech processing facility;

identifying by speech processing an utterance identifying a called party, said utterance transmitted via said first link to said speech processing facility;

selecting a second customer profile record corresponding to a virtual office equipment designation assigned to said called party;

replacing said first customer profile record with said second customer profile record;

pursuant to information in said second customer profile record establishing a speech connection between said first link and a message storage associated with said second link;

storing a voice message for said called party in said message storage.

12. A method according to claim 11 wherein said utterance identifying said called party emanates from a party initiating said request to make said voice call.

13. A method according to claim 11 including the step of transmitting said stored message responsive to speech processing identification of a party requesting said message.

14. A method according to claim 13 wherein said speech processing is responsive at least in part to information in said second customer profile record following reinstallation of said second customer profile record at said central office switching system serving said second link.

15. A method according to claim 14 wherein said reinstallation is responsive to processing of speech of said called party.

16. A method according to claim 15 wherein the speech of said called party which is processed is received by said processing facility via said second link.

17. A method according to claim 11 wherein said second link is associated with multiple customers, each of said multiple customers being assigned a personalized customer profile record corresponding to an individual virtual office equipment designation.

18. A method for processing a call in multilink telecommunication network comprising the steps of:

assigning subscriber profiles to subscribers associated with a first of said links connected to a first switching system;

at least one of said subscriber profiles including data indicative of subscription to messaging service;

designating in a messaging system a storage capability for said at least one subscriber profile;

identifying through speech signal processing the subscriber profile of one of said subscribers to whom an attempt is made to initiate a call via a second link;

processing said attempt to initiate said call in a manner based at least in part on processing information in the subscriber profile identified by said speech signal processing including said data indicative of subscription to messaging service;

establishing a speech link between said second link and said messaging system; and

storing in said designated storage capability in said messaging system a message transmitted via said second link.

19. A method according to claim 18 wherein said subscriber profile identified by said speech signal processing is selected from said subscriber profiles assigned to said multiple subscribers associated with said first of said links by a virtual equipment designation.

20. A method according to claim 18 including the steps of assigning virtual office equipment designations to each of said subscriber profiles, and selecting from said subscriber profiles said subscriber profile identified by said speech signal processing by its virtual equipment designation.

21. A method according to claim 20 including the steps of assigning to said first link an office equipment designation designating office equipment in a switching system in said multilink telecommunication network, which switching system serves said first link, said office equipment designation identifying a further customer profile, and identifying

connection is established to said speech signal processing facility, the identity of one of said subscribers associated with said second link is established via said speech signal processing facility, a virtual office equipment designation identifying a service profile of said one subscriber is associated with the switching system to which said second link is connected, pursuant at least partially to information in said service profile storing a voice message from said first link in said message storage facility for retrieval by said identified subscriber.

37. A communication network comprising:

multiple central office switching systems connected to multiple links, said links including a first and a second link, said second link having multiple subscribers associated therewith;

a speech signal processing facility; and

a message storage facility;

a signaling and control network connected to said central office switching systems, said speech signal processing facility; and said message storage facility;

wherein responsive to initiation of a request to establish a connection from said first to said second links, a connection is established to said speech signal processing facility, the identity of one of said subscribers associated with said second link is established, a virtual office equipment designation identifying a service profile of said one subscriber is associated with the switching system to which said second link is connected, pursuant at least partially to information in said service profile storing a voice message from said first link in said message storage facility for retrieval by said identified subscriber.

38. A communication network comprising:

multiple central office switching systems connected to multiple links, said links including a first and a second link, said second link having multiple subscribers associated therewith;

a speech signal processing facility;

a message storage facility;

a signaling and control network connected to said central office switching systems, said speech signal processing facility; and said message storage facility;

wherein responsive to initiation of a request to establish a connection from said first to said second links, a first service profile corresponding to said second link is installed in the central office switching system serving said second link and pursuant at least partially to information in said first service profile connection is established to said speech signal processing facility, the identity of one of said subscribers associated with said second link is established, a second service profile corresponding to said one subscriber is installed in said central office switching system serving said second link, pursuant at least partially to information in said second service profile a voice message from said first link is stored in message storage associated with said second link for retrieval by said one party.

39. A system according to claim 38 wherein said switching systems are connected in a telecommunications network.

40. A system according to claim 39 wherein responsive to detecting a request for service over said second link subsequent to storage of said message, said first service profile in installed in said central office switching system serving said second link, responsive to an utterance over said second link, said second service profile is substituted for said first service profile in said central office switching system serving said second link, and pursuant at least partially to information in said second service profile said stored voice message is transmitted via said second link.

41. A method comprising:

detecting a request to make a voice call from a first link to a second link through a communication network including multiple central office switching systems connected to multiple links;

installing in the central office switching system serving said second link a service profile generic to said second link;

pursuant to said generic service profile identifying one of multiple parties available via said second link by processing signals resulting from speech transmitted via said first link;

responsive to said identification installing in the central office switching system serving said second link a service profile specific said one party;

pursuant at least partially to information in said specific service profile storing a voice message from said first link in message storage associated with said second link and with said one party for retrieval by said one party.

* * * * *

US006101242A

United States Patent [19]

McAllister et al.

[11] **Patent Number:** 6,101,242

[45] **Date of Patent:** *Aug. 8, 2000

[54] MONITORING FOR KEY WORDS WITH SIV TO VALIDATE HOME INCARCERATION

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[73] Assignee: Bell Atlantic Network Services, Inc., Arlington, Va.

[*] Notice: This patent is subject to a terminal disclaimer.

[21] Appl. No.: 08/997,505

[22] Filed: Dec. 23, 1997

Related U.S. Application Data

[63] Continuation-in-part of application No. 08/828,959, Mar. 28, 1997, Pat. No. 5,978,450, and a continuation-in-part of application No. 08/904,936, Aug. 1, 1997.

[51] Int. Cl.[7] H04M 7/00

[52] U.S. Cl. 379/88.02; 379/201; 379/207

[58] Field of Search 379/88.01, 88.02, 379/88.03, 201, 207, 210, 211, 212, 216, 229, 230, 142, 38

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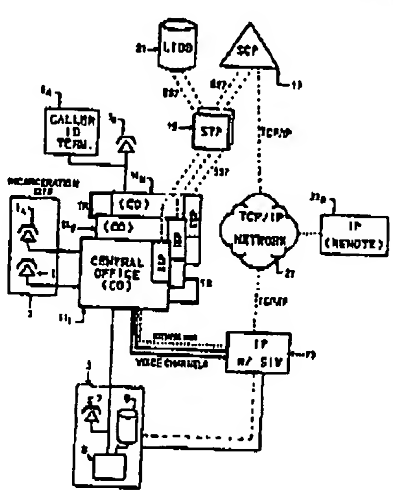
Primary Examiner—Krista Zele
Assistant Examiner—Roland G. Foster
Attorney, Agent, or Firm—McDermott, Will & Emery

[57] ABSTRACT

A telephone network provides personalized services based on voice identification of a person making or receiving a call. For example, when a person initiates a call, the network executes a speaker identification/verification procedure to identify the person as a subscriber. The identification/verification process provides a virtual office equipment number corresponding to the identified subscriber. The central office switch servicing the outgoing call receives this virtual number and uses it to retrieve a service profile associated with the subscriber. The switch provides a personalized telephone service to the subscriber during processing of the call, using the retrieved service profile. The personalized service may include a number of unique features, such as recording call related data together with data about the identification of the person. For home incarceration or the like, the service may entail monitoring speech communications to detect certain designated words and automatically terminate the call.

48 Claims, 7 Drawing Sheets



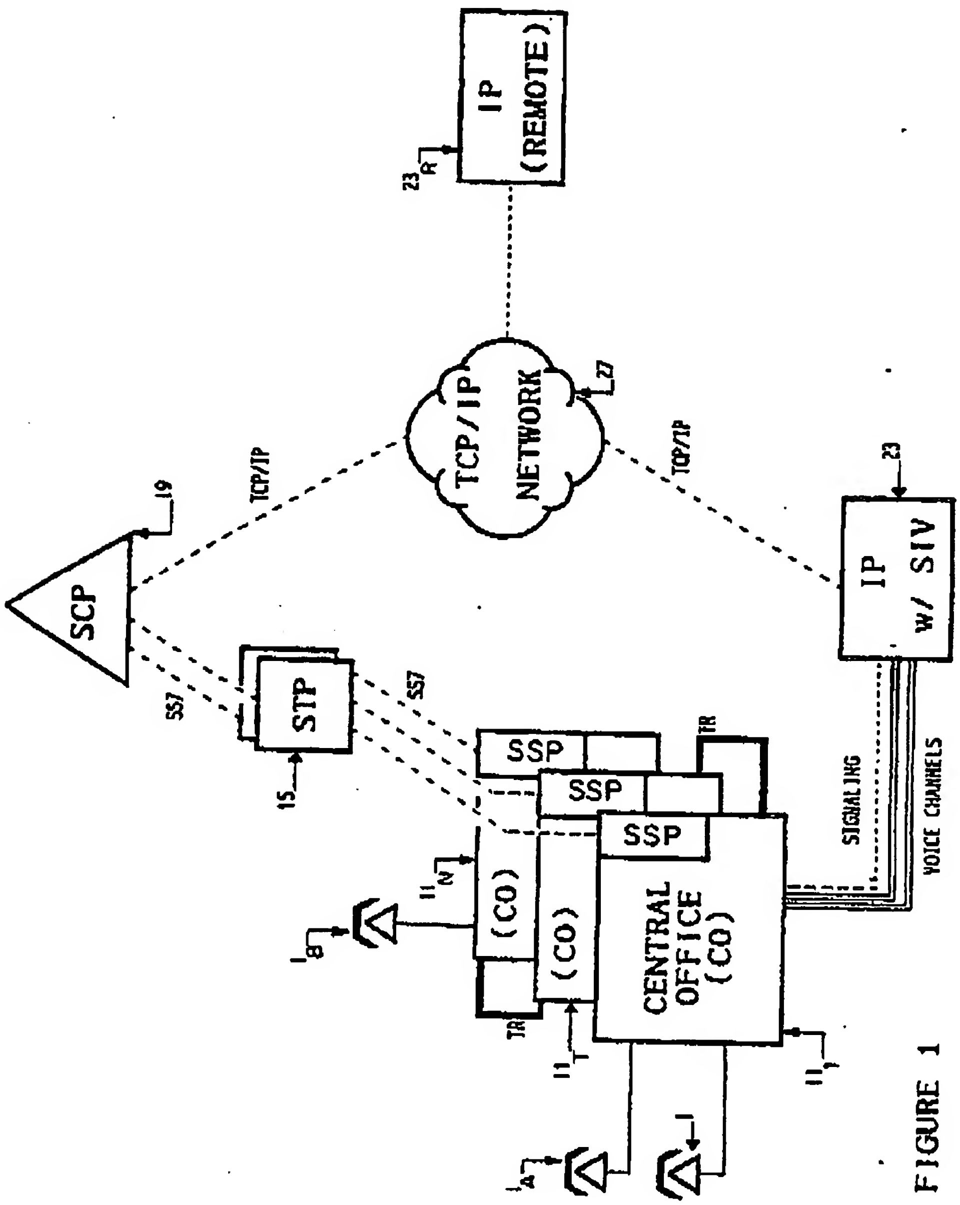

IP
(REMOTE)
23R
TCP/IP
NETWORK
27
TCP/IP
TCP/IP
SCP
19
SS7
STP
15
SS7
SSP
SSP
SSP
TR
TR
CENTRAL
OFFICE
(CO)
(CO)
(CO)
IP
w/ SIV
23
SIGNALING
VOICE CHANNELS

FIGURE 1

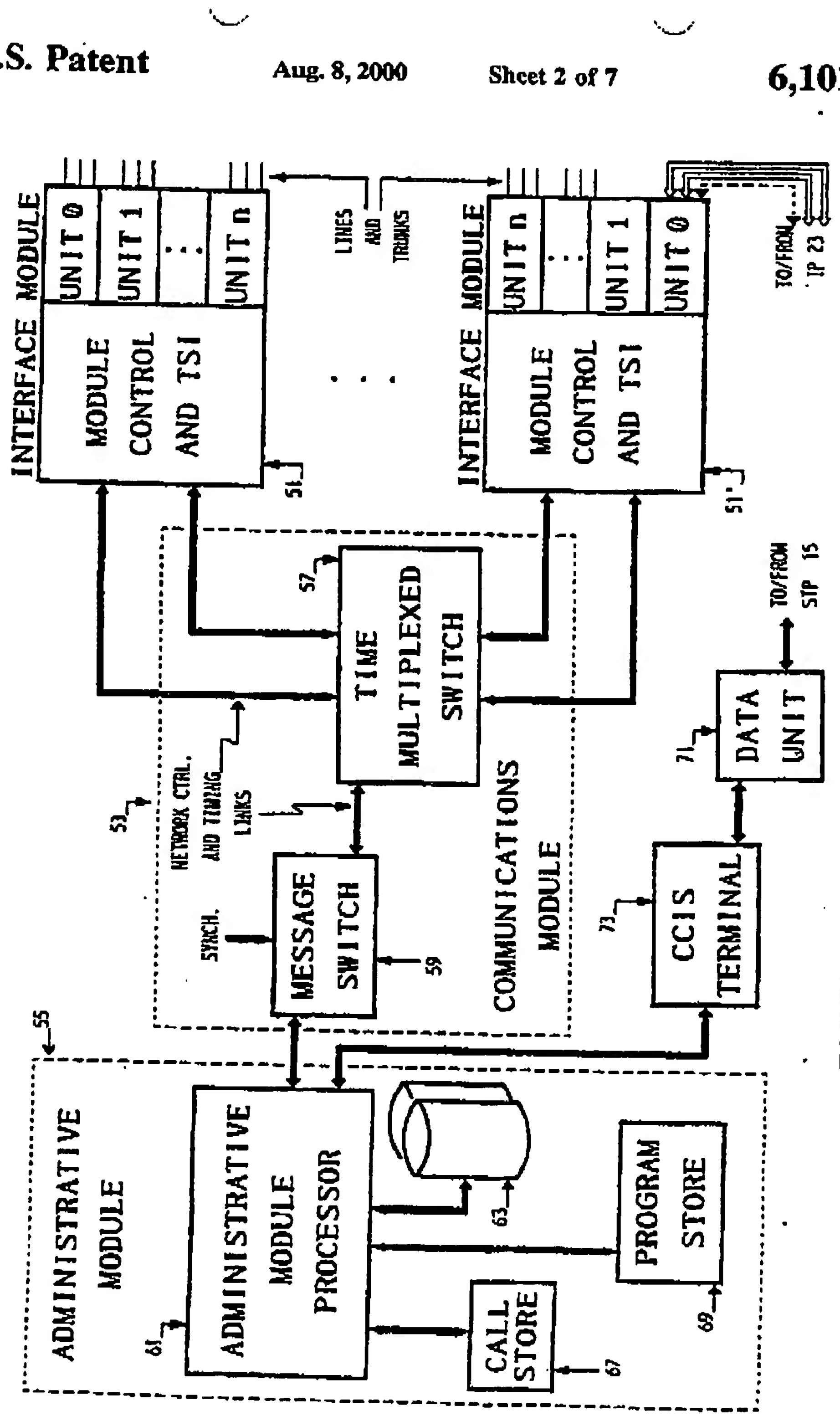
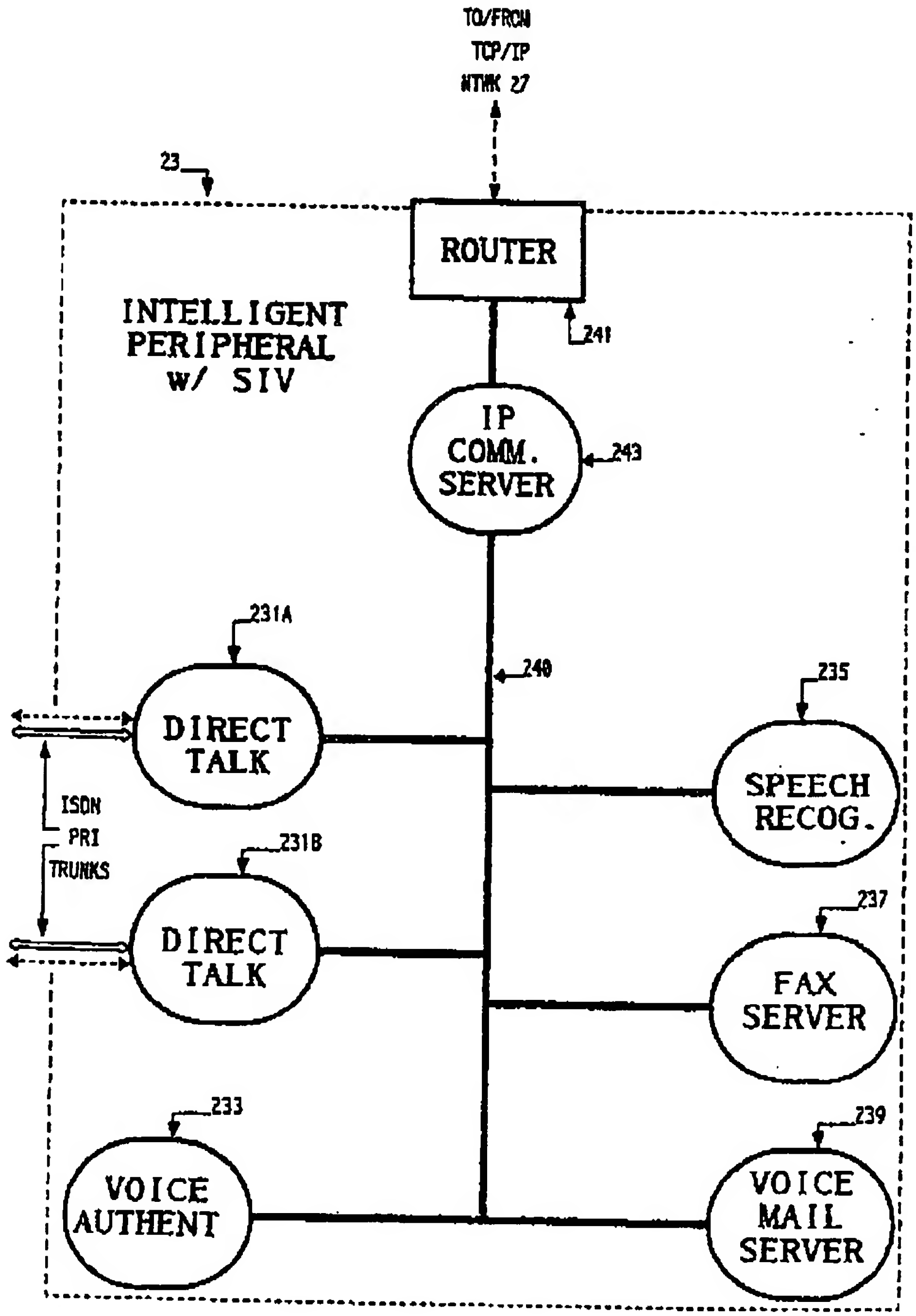

INTERFACE MODULE
UNIT 0
UNIT 1
UNIT n
MODULE CONTROL AND TSI
51
LINES AND TRUNKS
INTERFACE MODULE
UNIT n
UNIT 1
UNIT 0
MODULE CONTROL AND TSI
51'
TO/FROM IP 23
57
TIME MULTIPLEXED SWITCH
NETWORK CTRL. AND TIMING LINKS
53
SYNCH.
MESSAGE SWITCH
59
COMMUNICATIONS MODULE
TO/FROM STP 15
71
DATA UNIT
73
CCIS TERMINAL
55
ADMINISTRATIVE MODULE
61
ADMINISTRATIVE MODULE PROCESSOR
63
PROGRAM STORE
69
CALL STORE
67

FIGURE 2



TO/FROM TCP/IP NTWK 27
23
ROUTER
241
INTELLIGENT PERIPHERAL w/ SIV
IP COMM. SERVER
243
231A
240
235
DIRECT TALK
SPEECH RECOG.
ISDN PRI TRUNKS
231B
237
DIRECT TALK
FAX SERVER
233
239
VOICE AUTHENT
VOICE MAIL SERVER

FIGURE 3

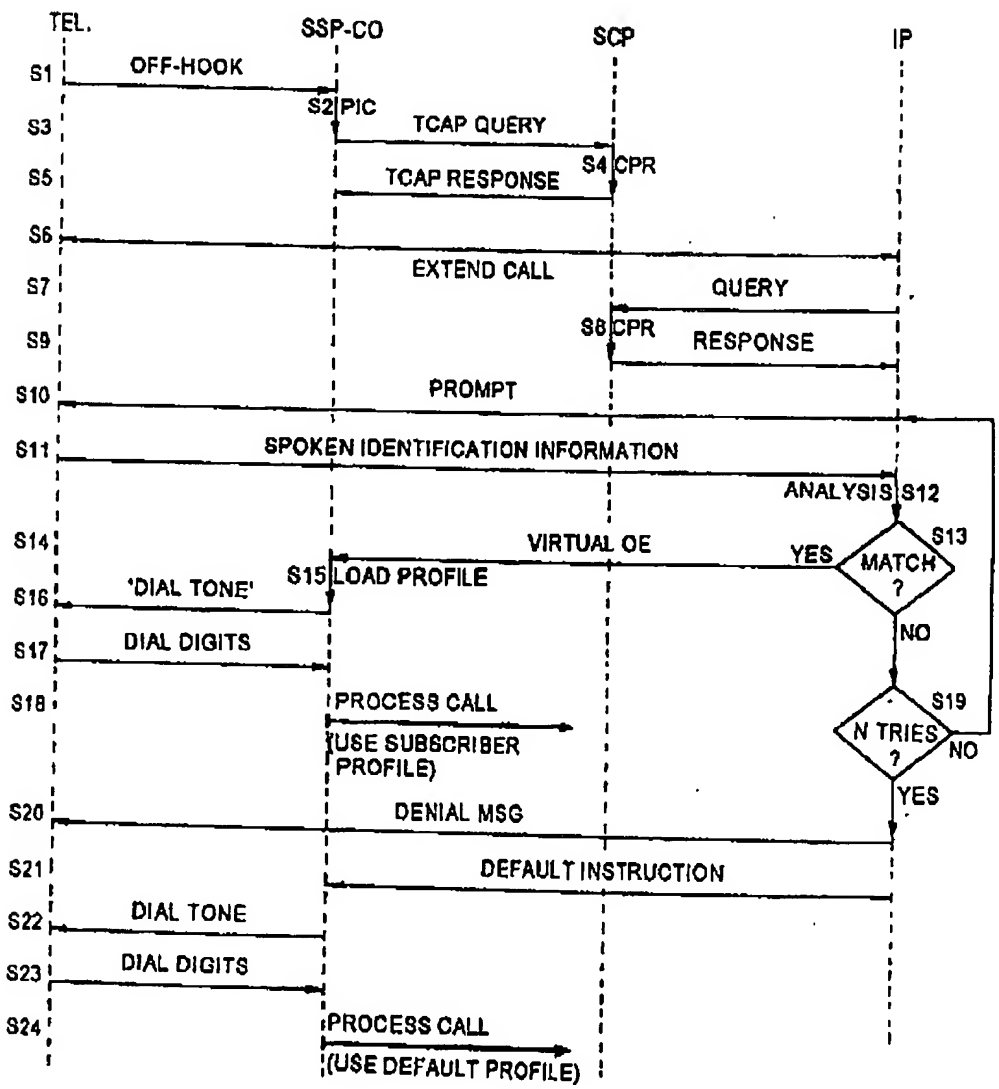
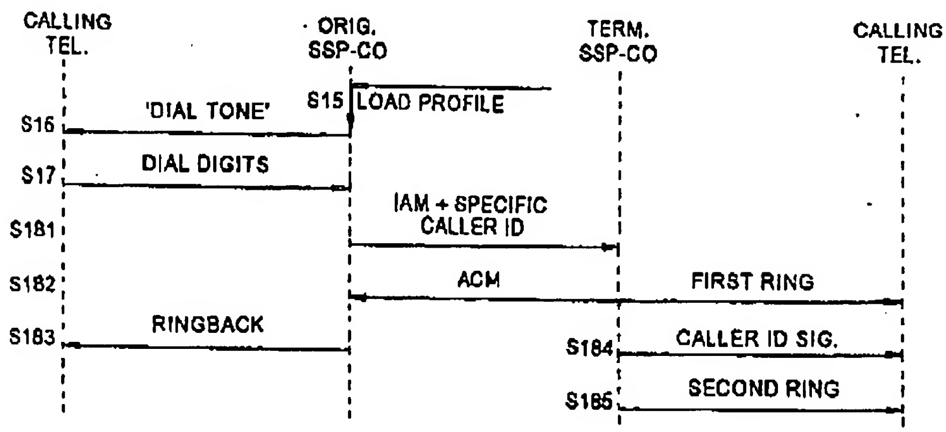
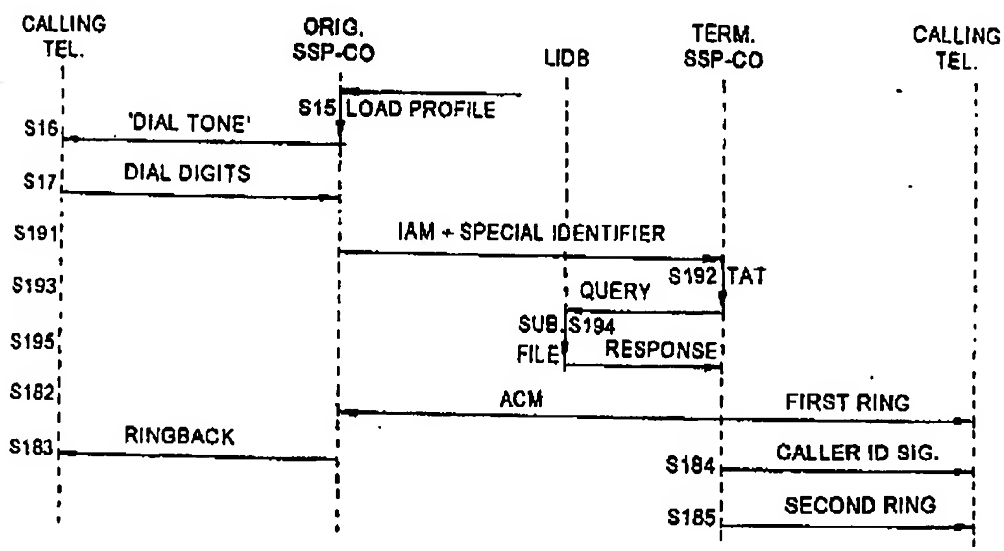

TEL.
SSP-CO
SCP
IP
S1
OFF-HOOK
S2 PIC
S3
TCAP QUERY
S4 CPR
S5
TCAP RESPONSE
S6
EXTEND CALL
S7
QUERY
S8 CPR
S9
RESPONSE
S10
PROMPT
S11
SPOKEN IDENTIFICATION INFORMATION
ANALYSIS S12
S13
S14
VIRTUAL OE
YES
MATCH ?
S15 LOAD PROFILE
S16
'DIAL TONE'
NO
S17
DIAL DIGITS
S18
PROCESS CALL (USE SUBSCRIBER PROFILE)
S19
N TRIES ?
NO
YES
S20
DENIAL MSG
S21
DEFAULT INSTRUCTION
S22
DIAL TONE
S23
DIAL DIGITS
S24
PROCESS CALL (USE DEFAULT PROFILE)

FIG. 4A



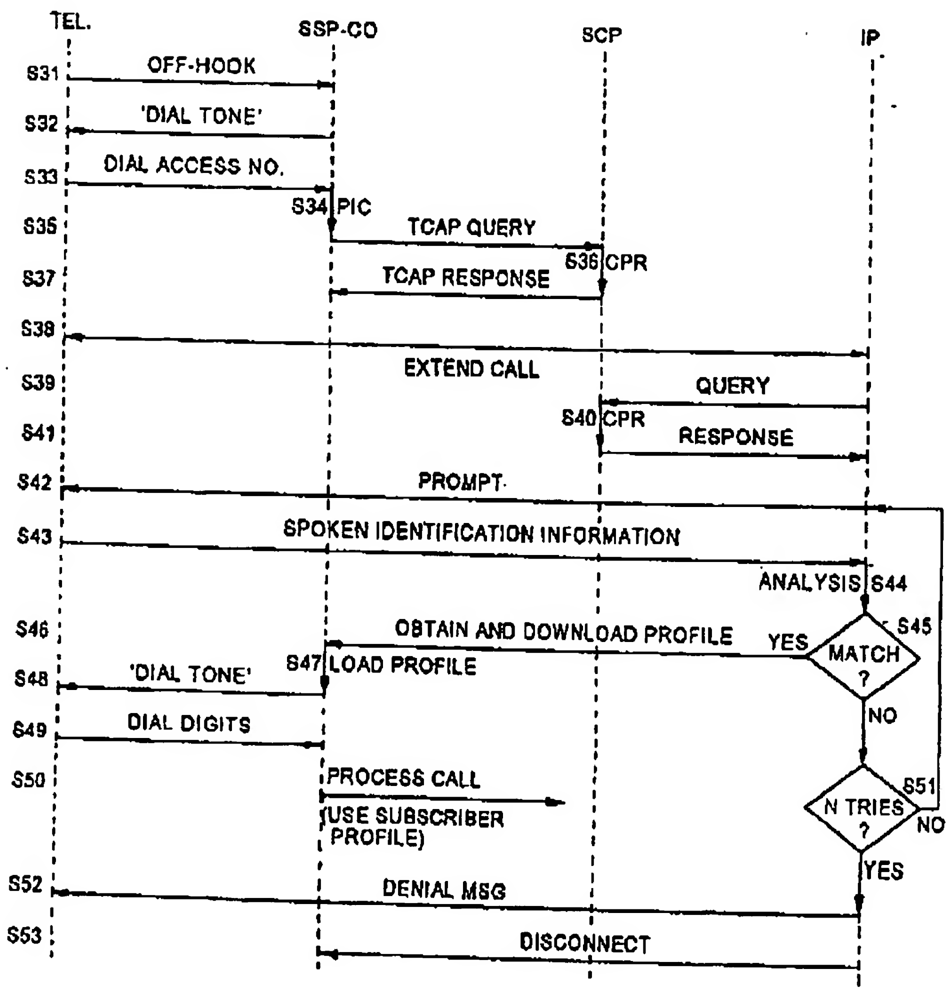
CALLING TEL.
ORIG. SSP-CO
TERM. SSP-CO
CALLING TEL.
S15
LOAD PROFILE
S16
'DIAL TONE'
S17
DIAL DIGITS
S181
IAM + SPECIFIC CALLER ID
S182
ACM
FIRST RING
S183
RINGBACK
S184
CALLER ID SIG.
S185
SECOND RING

FIG. 4B



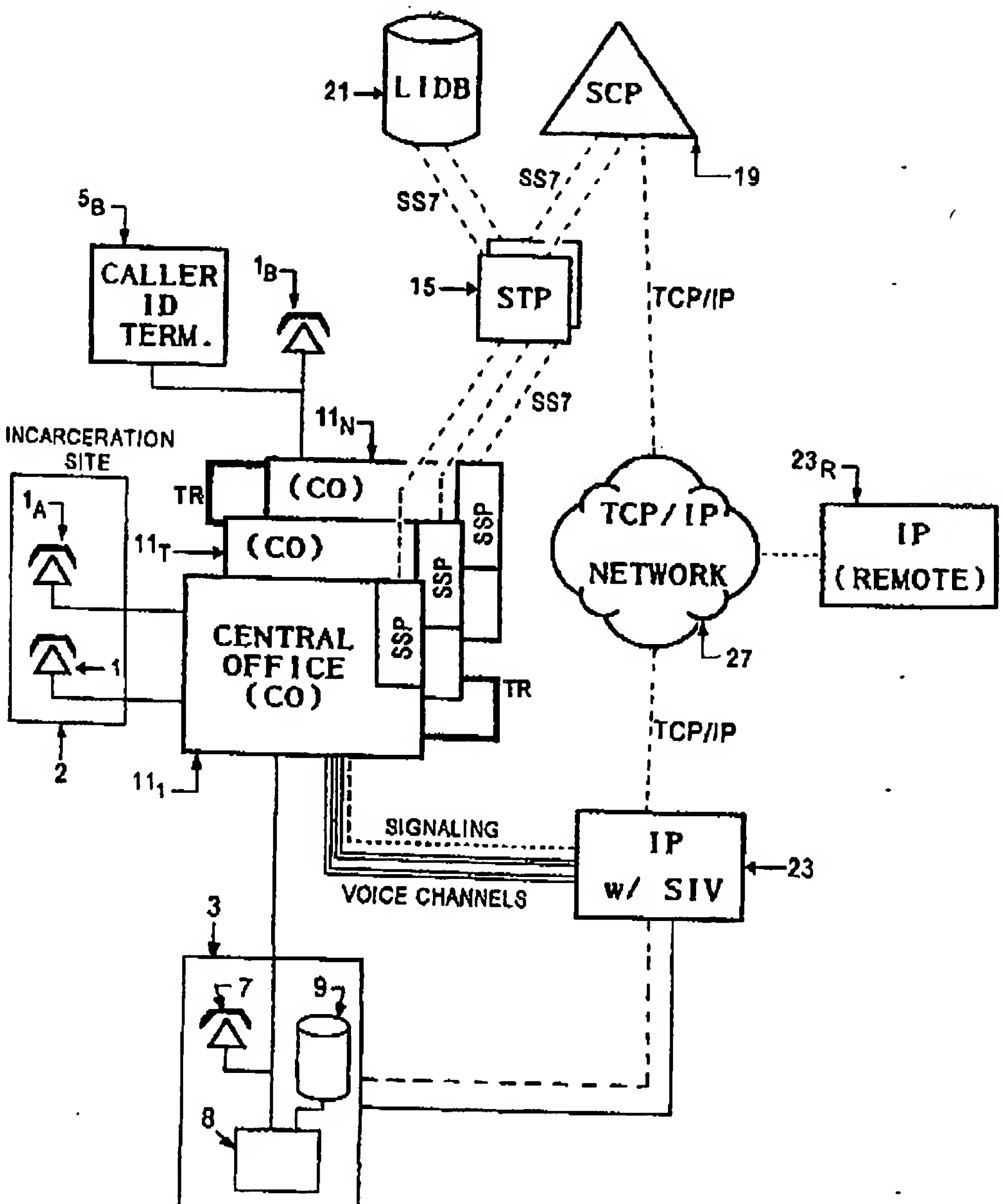
CALLING TEL.
ORIG. SSP-CO
LIDB
TERM. SSP-CO
CALLING TEL.
S15
LOAD PROFILE
S16
'DIAL TONE'
S17
DIAL DIGITS
S191
IAM + SPECIAL IDENTIFIER
S192
TAT
S193
QUERY
SUB. FILE
S194
S195
RESPONSE
S182
ACM
FIRST RING
S183
RINGBACK
S184
CALLER ID SIG.
S185
SECOND RING

FIG. 4C



TEL.
SSP-CO
SCP
IP
S31
OFF-HOOK
S32
'DIAL TONE'
S33
DIAL ACCESS NO.
S34
PIC
S35
TCAP QUERY
S36
CPR
S37
TCAP RESPONSE
S38
EXTEND CALL
S39
QUERY
S40
CPR
S41
RESPONSE
S42
PROMPT
S43
SPOKEN IDENTIFICATION INFORMATION
ANALYSIS
S44
S46
OBTAIN AND DOWNLOAD PROFILE
YES
MATCH ?
S45
S47
LOAD PROFILE
S48
'DIAL TONE'
NO
S49
DIAL DIGITS
S50
PROCESS CALL
(USE SUBSCRIBER PROFILE)
S51
N TRIES ?
NO
YES
S52
DENIAL MSG
S53
DISCONNECT

FIG. 5



21
LIDB
SCP
SS7
SS7
19
5B
CALLER ID TERM.
1B
15
STP
TCP/IP
SS7
11N
INCARCERATION SITE
23R
TR
(CO)
SSP
TCP/IP NETWORK
IP (REMOTE)
1A
11T
(CO)
SSP
CENTRAL OFFICE (CO)
SSP
1
27
TR
2
11₁
TCP/IP
SIGNALING
IP w/ SIV
23
VOICE CHANNELS
3
7
9
8

FIG. 6

MONITORING FOR KEY WORDS WITH SIV TO VALIDATE HOME INCARCERATION

CROSS-REFERENCE TO RELATED APPLICATION

This application is a continuation-in-part of U.S. patent application Ser. No. 08/828,959 filed Mar. 28, 1997 (now U.S. Pat. No. 5,978,450), and of U.S. patent application Ser. No. 08/904,936 filed Aug. 1, 1997, the disclosures of which are incorporated herein entirely by reference.

TECHNICAL FIELD

The present invention relates to personalized telecommunications service, preferably offered through an intelligent telephone network. In particular, the present invention relates to speaker identification verification and monitoring for key words to validate and control communication in home incarceration.

ACRONYMS

The written description uses a large number of acronyms to refer to various services, messages and system components. Although generally known, use of several of these acronyms is not strictly standardized in the art. For purposes of this discussion, acronyms therefore will be defined as follows:

Address Complete Message (ACM)
Advanced Intelligent Network (AIN)
ANswer Message (ANM)
Automatic Number Identification (ANI)
Call Processing Record (CPR)
Central Office (CO)
Common Channel Interoffice Signalling (CCIS)
Data and Reporting System (DRS)
Destination Point Code (DPC)
Generic Data Interface (GDI)
Initial Address Message (IAM)
Integrated Service Control Point (ISCP)
Integrated Services Digital Network (ISDN)
ISDN User Part (ISDN-UP)
Intelligent Peripheral (IP)
Line Identification Data Base (LIDB)
Multi-Services Application Platform (MSAP)
Office Equipment (OE)
Origination Point Code (OPC)
Personal Communications Service (PCS)
Plain Old Telephone Service (POTS)
Point in Call (PIC)
Personal Identification Number (PIN)
Primary Rate Interface (PRI)
Public Switched Telephone Network (PSTN)
Service Control Point (SCP)
Service Creation Environment (SCE)
Service Management System (SMS)
Service Switching Point (SSP)
Signaling System 7 (SS7)
Signaling Point (SP)
Signaling Transfer Point (STP)
Simplified Message Desk Interface (SMDI)
Speaker Identification/Verification (SIV)
Terminating Attempt Trigger (TAT)
Time Slot Interchange (TSI)
Traffic Service Position System (TSPS)
Transaction Capabilities Applications Part (TCAP)
Transmission Control Protocol/Internet Protocol (TCP/IP)

BACKGROUND ART

The concept of home incarceration has evolved as an alternative to detention in government jail and prison facili-

ties. In cases of relatively light infractions, offenders, rather than being placed as inmates in overcrowded facilities, are confined to predetermined limited geographical areas including, for example, homes, so called half-way houses, workplaces, and the like. The burden on the prison system is relieved by creating additional space for criminals convicted of more serious crimes. Cost efficiency is also a significant factor as the expense of incarceration in a high security facility is quite high. Additionally, the degree of severity of punishment and the prospects of rehabilitation of the light offender are more appropriate to a home incarceration environment than in a prison provided for felons.

In a "house arrest" situation, the detainees, are more likely to have a certain degree of interaction with the community. Public security is a socially sensitive and obviously critical issue and it is important that the activities of such captives be monitored and supervised. The whereabouts and identity of individuals should be capable of being established at any time without the necessity of assignment of a law enforcement officer for constant surveillance on a one to one basis.

U.S. Pat. No. 5,170,426, issued Dec. 8, 1992 to the assignee of the instant application discloses one method and system for home incarceration. According to that system monitoring and verification is performed through a telephone network including a telephone on the premises of the location of confinement and a control center. Voice verification, using voice analysis of speech transmitted in a telephone call from the site to the center is performed during periodic testing. A voice template vocabulary is established for the individual and used for voice verification. Caller line identification of each incoming call is performed to verify that call originates from the appropriate location. The confined individual is required, either randomly or at scheduled intervals, to call the control center and recite a statement including randomly selected words from the template vocabulary.

U.S. Pat. No. 4,843,377 shows an arrangement for home incarceration which contemplates the use of a voiceprint as a means for remote prisoner identification. Audio spectral analysis is performed and applied to speech transmitted over a telephone line to determine a match with a probationer's voiceprint. Several commercially available systems are discussed.

Conventional home incarceration systems generally require one line and associated telephone and office equipment number (OE) for each incarcerated person. This arrangement is relatively costly and requires the use of a large number of individual telephone numbers, which are now in diminishing supply in the current numbering plan.

The public switched telephone network (PSTN) and other telephone networks, such as cellular systems, provide most telephone services based on number identification of the telephone set or line that each party uses. The services which are enabled are individualized only to the extent that a party uses the same line and/or instrument. For example, a customer is typically provided with one set of service features and billing options available via a telephone on an office desk, another set of service features and billing options available via a telephone line to the home, and perhaps a third set of service features and billing options available via a wireless telephone (e.g. cellular or personal communications service (PCS)).

The networks process calls to and from each of these different subscriber telephones based on a separate telephone number. Also, a caller may use personalized billing options by using a calling card, but often the input opera-

tions for calling card service are overly complex. With the exception of calling card billing, each person using a particular telephone typically can access only those service features and billing options which are associated with the particular line or telephone instrument.

This phenomenon has already caused subscribers inconvenience in the proliferation of services now being proposed or implemented. For example, circumstances arise in which a subscriber may want a feature or billing option normally associated with one line or instrument, such as the office telephone, when they are in fact using a different line or instrument, such as their home or PCS telephone. Alternatively, two or more persons using one telephone or line often want different sets of service options. The extreme increase in demand for telephone services is rapidly exhausting the capacity of the network, particularly in terms of the telephone numbers available under the current numbering plan.

A number of specific solutions have been proposed for specific problems, such as work at home and/or transfer of service to new location(s) as an individual travels. However, each of these solutions is limited or creates its own new problems.

For example, U.S. Pat. No. 4,313,035 to Jordan et al. discloses a method of using an intelligent network to provide a 'follow-me' type service through multiple exchanges of the switched telephone network using an AIN type of telephone system architecture. Each subscriber to the locator service has a unique person locator telephone number. To access the system to update data in a service control database, the subscriber dials 0700 and his unique person locator telephone number. The telephone switching office routes the call to a traffic service position system (TSPS) which prompts the caller (e.g. provides an additional dial tone) and receives further digits from the subscriber. The subscriber inputs a three digit access code, indicating the type of update call, and a four digit personal identification number. If calling from the remote station to which the subscriber wishes his calls routed, the local switching office forwards the line identification number of that station to the TSPS. The TSPS forwards the dialed information and the line identification to the data base for updating the particular subscriber's location record. A caller wishing to reach the subscriber dials the subscriber's unique person locator number. A telephone switching office sends the dialed number to the central database. The database retrieves the stored completion number for the called subscriber and forwards that number back to the switching office to complete the call.

The Jordan et al. approach allows calls to follow the subscriber to each new location, but the subscriber must have a unique telephone number for this service. Each station that receives a call also must have a unique telephone number. As such, the Jordan et al. approach actually exacerbates the shortage of telephone numbers. Also, Jordan et al. rely on subscriber input of identification numbers. Subscribers often find this inconvenient, and this technique is often prone to number entry errors.

U.S. Pat. No. 4,899,373 to Lee et al. discloses a system for providing special telephone services to a customer on a personal basis, when the customer is away from his or her home base or office. The personalized services are provided in a multiple exchange office environment, using a central database for feature control. The nationally accessible central database system stores feature data in association with personal identification numbers. A subscriber wishing to use his personalized features while away from home base dials a special code and presents the personal identification number. The exchange transmits a query to the central database, and the corresponding feature data is retrieved from the database. The database forwards the feature data to the exchange, and the exchange stores the received feature data in association with the station from which the request was initiated. Subsequently, the exchange accesses the downloaded feature data to provide telephone service corresponding to the subscriber's personalized telephone features via the station the subscriber is currently operating from. A temporary office arrangement may be established in which the personalized features will be immediately available on incoming and outgoing calls for a period of time specified by the subscriber.

U.S. Pat. No. 5,206,899 to Gupta et al. pertains to a system wherein a subscriber can assign desired characteristics to any "target station" which is an active telephone accessible to a telecommunications network. A call thereafter that originates from the target station can use customized features, such as account code dialing and corporate billing arrangements. Initially, a service profile is created and stored for each subscriber and contains information describing desired features and billing options. The characteristics of a particular target station are changed by an activation process that can be initiated from any location. Automatic number identification (ANI) information associated with the target station is entered into an ANI trigger table in an intelligent switch, and the service profile is loaded into a database. When a call originates from the target station, information in the database is applied to the switch to provide the desired characteristics. An example of one of the features is when an employee of company X wishes to make business related calls from his/her telephone, the call has the characteristics of a call made from the office by a special billing arrangement.

Like Jordan, the Lee et al. and Gupta et al. systems depend on a dialed number entry by the subscriber to activate the service. Also, the Lee et al. and Gupta et al. systems do not provide a simple manner for more than one subscriber to obtain personalized service over the same telephone line. In Lee et al., during the period when the switch stores the roaming subscriber's profile in association with the line, all calls are processed based on that one profile. Similarly, in Gupta et al., while the ANI trigger is set against the line, all outgoing calls cause database access and use of the subscriber's profile in the database. There is no way to fall back on the normal profile for that line unless and until the service for the roaming subscriber is cancelled with respect to that one line.

U.S. Pat. No. 5,247,571 to Kay et al. discloses an Area Wide Centrex service provided by an advanced intelligent telephone network. The service provides centrex features, such as extension dialing, to multiple locations. The Kay et al. Patent also suggests a Work-at-Home feature. This feature allows the home telephone line to selectively operate as a residential line or as a Centrex business line, on a call-by-call basis. For a business call, the user would preface each call with an access indicator to identify a business call. When an outgoing call from the home line lacks the access indicator, the network processes the call as a standard residential call.

The Work-at-Home feature in the Kay et al. system requires only dialing of a code before each outgoing business call. However, the Kay et al. approach requires that the business profile is stored in association with the home line before the subscriber makes the call. The subscriber can use the Centrex billing and service features from the business

account only from a home telephone previously associated with the business line. The subscriber can not use the billing and service features from the business account from any randomly selected telephone. Also, from the home line, a person can either use the normal residential profile service or the pre-defined business profile service. There is insufficient flexibility to enable a wider range of services for multiple subscribers through the one line.

An increasingly popular telephone service is caller identification or 'caller ID'. The telephone network identifies the telephone number associated with the line or instrument used by the calling party and supplies the number and or the name to a display device at the called customer's premises. This procedure, or the ANI (automatic number identification) equivalent thereof, has been relied upon in prior home incarceration services to identify the line assigned to each detainee.

Subscribers having ISDN service receive caller ID data, for display at the time of an incoming call, in the form of a data message which the end office switch transmits over the D-channel. For analog telephone customers, however, existing caller ID utilizes in-band transmission technology similar to that described in U.S. Pat. Nos. 4,582,956 and 4,551,581 to Doughty. In such a system, the end office switch connected to the called party's line transmits directory number data for the calling party's telephone line as frequency shift keyed (FSK) data inserted in the silent interval between ringing signal pulses applied to the called party's line. The receiving apparatus includes a line interface unit, a converter, a control circuit and a display unit. A frequency shift keyed (FSK) signal representing the special service information is filtered from the ringing signals by the line interface unit. The converter detects the FSK signal and demodulates the special service information from the FSK signal. Following detection of the FSK signal, the control circuit receives and stores the special service information. The stored information is periodically sent to the display unit to begin exhibiting thereof during the silent interval before the next ringing signal.

The local telephone exchange carriers have recently begun offering an enhanced form of caller ID, sometimes referred to as 'Caller ID Deluxe' service. This enhanced service utilizes AIN type call processing to access a Line Information Database (LIDB) to translate the calling party's directory number into name data. The end office switch forwards the name data and the normal caller ID telephone number as FSK encoded data inserted in the silent intervals between ringing signals.

The LIDB database includes a single listing for each telephone line and translates each number into a single name, typically the name of the party identified as the customer or subscriber for billing purposes. In fact, the LIDB database provides this single translation even for calls from one line having multiple telephone numbers. Consider an example in which a family has one line with two numbers and a distinctive ringing service. The first number is used for the family as a whole, and the second number is used for a teenage son or daughter. The distinctive ringing allows people in the household to know whether or not each call is for the teenager. On outgoing calls, however, the end office switch always identifies the line by the primary number (the family's number), and the LIDB database always provides the name of the billing subscriber, e.g. the father's name. As a result, when the teenager calls a friend, the friend will receive the main number and possibly the father's name. If the friend calls back using the information from his caller ID display terminal, the friend calls the family's main number, not the teenager's number.

However, the above discussed examples of prior suggestions to customize services have not adapted the caller identification to correspond to the actual party using the telephone on the outgoing call. For example, in a system like that of Lee, Gupta or Kay, the caller might use features and billing options associated with her personalized or work service, but any such calls would produce a caller ID display identifying the number of the station from which she originated the call. If the called party subscribed to the name type enhanced caller ID, the network would provide a name associated with that telephone number, not the name of the actual calling party.

U.S. Pat. Nos. 4,961,217 and 4,759,056 disclose a card based system for providing personalized features, including caller name display. Each user has a "portable memory device" in the form of an identification card bearing personal information including identification information. When initiating a call, the user inserts the card in the calling station, and information from the card is transmitted to the central switching system. In one embodiment, the switching system translates the identification information from the card to produce a textual representation of the calling party's name and transmits that information to a called terminal for display. Although this system does provide a name display identifying the actual called party, the system requires the use of the identification card and specialized calling terminals for reading the information from the cards.

As shown by the above discussion, a need still exists for an effective and convenient system for providing individualized calling service features, including actual caller as well as line identification.

DISCLOSURE OF THE INVENTION

The present invention addresses the above noted problems and provides advances over the existing technology by individualizing telecommunication services based on a speech authenticated identification of the actual calling person as well as the actual answering person. Offices of a communication network utilize profile data associated with an identified person, rather than profile data associated with a particular telephone number or a particular communication link. In some of the preferred service embodiments, the network uses a virtual office equipment number assigned to the person's profile data to retrieve the data for providing a requested service, such as home incarceration, reducing or eliminating the need for assignment of additional telephone numbers. The network also provides caller identification information to a called destination, including data, such as an identity by name and voice, specifically associated with the person speaking into the telephone instrument.

Thus, in one aspect the present invention relates to a method of providing service through a communication network. An attempt to make a call from a predetermined link through the network is detected. The next step in the method is receiving and processing speech signals from a person via the predetermined link. The processing identifies the person making the call as one of a number of individuals expected to seek services offered through the communication network. An instruction is sent to a switching office of the network instructing that office to utilize profile data corresponding to the identified individual for processing of the call. As part of subsequent call processing, the network transmits data representing the identity of the subscriber to a predetermined destination. In the home incarceration embodiments, the data may be the identity of the person speaking into the telephone instrument by name or a short

code identifying by voice recognition the person as one of the individuals associated with the station or line from which the person initiated or answered the call.

Another aspect of the invention relates to a call processing methodology using profile information selected by means of a virtual office equipment number. This method includes identifying one party utilizing a specific communication service, for example the person making an outgoing call, as one of a plurality of persons. Using a virtual office equipment number, assigned to the identified one person, corresponding profile data is retrieved from storage. A communication network provides the requested communication service over a communication link, based at least in part on the retrieved profile data. As part of the service, a portion of the retrieved profile data is used to provide an identification of the one person over another link of the communication network.

In the preferred embodiments, the identification information or information derived therefrom is transmitted to a predetermined destination or destinations to identify either the person placing or the person answering the call to another participant to the communication service, e.g. to identify a specific incarcerated person as the person on the call. When transmitted to the destination, the data may be a code corresponding to the incarcerated person, but where the destination includes appropriate equipment, the data sent to the destination includes the name and other criteria specific to the incarcerated person. Several techniques are envisioned for providing the identifying data to the destination. In one embodiment, the incarcerated person profile data supplies all identifying criteria sought. In another embodiment, the profile data provides an identifier, and a remote database is accessed to translate the identifier into additional data.

Other aspects of the invention relate to the communication network implementing the individualized functions, including caller person specific identification. The preferred implementation of the communication network is an intelligent implementation of a public switched telephone network. The preferred network includes a number of central office switches interconnected by trunk circuits and serving a substantial number of telephone links. The intelligent network also includes a service control point storing a database of records used in controlling services provided through the central offices. A first signaling network carries signaling messages between the offices as well as signaling messages between the offices and the service control point. The peripheral also may exchange signaling information with the service control point, preferably over a second signaling network.

Another aspect of the invention relates to an improved central office switching system capable of processing a call using profile information selected in response to a virtual equipment number. An office equipment number is 'virtual' where it is assigned to an individual person, instead of to specific network equipment such as a line termination or a specific station.

The switching system includes interface modules coupled to the communication links and a switch providing selective communication connections between the interface modules. An administrative module controls connections provided by the switch. The administrative module includes mass storage containing subscriber profiles, a processor for providing control instructions to the switch, and a signaling interface for signaling communication with at least one external network node. In response to a virtual office equipment number received via the signaling interface, e.g. from a separate peripheral platform as discussed above, the processor retrieves a subscriber profile corresponding to the virtual office equipment number from the mass storage. The processor uses the retrieved profile to process a selective connection through the switch between two of the interface modules. In accord with this aspect of the invention, the retrieved profile data actually provides data identifying the person, typically for use in identifying the person to an incarceration supervising authority or agent.

Advantages of the personal identification service should be readily apparent to those skilled in the telecommunications art. For example, in the shared line application, several incarcerated individuals can share a single line or communication link as well as a single telephone number. Outgoing call features, however, are personalized to each detainee. For example, the network can provide each subscriber a specifically individualized service, and each individual can receive a separate bill. The associated caller identification service provides an identification of each actual caller, not just one party named as the primary subscriber on one billing account for the originating station or line. The service uses speech based identification, eliminating the burden of dialing long strings of identifying digits. The system provides the ability to signal for a specific individual to answer, the ability to confirm that that individual did answer, the ability to confirm that the call was made, and the ability to associate the call with a charge or bill.

Additional objects, advantages and novel features of the invention will be set forth in part in the description which follows, and in part will become apparent to those skilled in the art upon examination of the following or may be learned by practice of the invention. The objects and advantages of the invention may be realized and attained by means of the instrumentalities and combinations particularly pointed out in the appended claims.

BRIEF DESCRIPTION OF DRAWINGS

The drawing figures depict the present invention by way of example, not by way of limitation. In the figures, like reference numerals refer to the same or similar elements.

FIG. 1 is a simplified block diagram of an intelligent telephone network that may be used to offer a personal dial tone service which may be utilized to provide one preferred embodiment of the present invention.

FIG. 2 is a simplified block diagram illustrating the significant functional components of an SSP type central office switching system used in the network of FIG. 1.

FIG. 3 is a simplified block diagram illustrating the significant functional components of an Intelligent Peripheral (IP) used in the network of FIG. 1.

FIG. 4A is a combination signal flow and process flow diagram useful in understanding a specific example of call processing for providing the personal dial tone service over a shared use line.

FIGS. 4B is a combination signal flow and process flow diagram useful in understanding a first embodiment of the processing for providing the identity of the actual caller to the destination display as caller ID information.

FIGS. 4C is a combination signal flow and process flow diagram useful in understanding another embodiment of the processing for providing the identity of the actual caller to the destination display as caller ID information.

FIG. 5 is a combination signal flow and process flow diagram useful in understanding a specific example of call

processing for providing the personal dial tone service on a dial-up, per call basis.

FIG. 6 shows a block diagram of a preferred embodiment of the telephone network shown in FIG. 1, which may be utilized to provide the home incarceration service of the invention.

BEST MODE FOR CARRYING OUT THE INVENTION

In response to each of several types of service requests, the personal dial tone service of the present invention initially identifies a calling or responding individual, who may be an incarcerated person or, in the more general application described in the above identified parent applications, the subscriber. This identification is established preferably by using a speaker identification/verification procedure. The system then retrieves profile information corresponding to the identified individual. The communication network processes one or more calls to or from an identified communication link using the individual's profile data.

On an outgoing telephone call from the individual, for example, the service request may be an off-hook signal, and the network may provide 'dial-tone' type telephone services based on the retrieved profile information. In this example, the network may provide a dial tone signal or a customized prompt and then permit the caller to out-dial a call. Caller identification, calling features and/or billing functions apply based on the profile information, e.g. to provide the name of the actual calling party, to bill the call to this one individual's personal account, and to prevent calls which are proscribed for an individual detainee. As will be described in further detail hereinafter, other specialized procedures may be called for in the profile of an incarcerated person, such as, for example, a monitoring for pre-specified key words. The network may also provide personalized services on incoming calls to the detainee.

The personal dial tone service in its broadest form may utilize a variety of different networks. For example, the service may be adaptable to Internet based voice communications. However, in the present home incarceration application such difficult to control applications as Internet applications would typically be prohibited. The preferred embodiments, including home incarceration, utilize various implementations of modern telephone networks. To understand the invention, it may be helpful first to consider the architecture and operation of an advanced intelligent network (AIN) type implementation of a public switched telephone network.

FIG. 1 provides a simplified illustration of the preferred intelligent telephone network for implementing the personal dial tone service in accord with the present invention. As shown, the telephone network includes a switched traffic network and a common channel signaling network carrying the control signaling messages for the switched telephone traffic network. In this implementation, the system further includes a secondary signaling network.

The telephone or traffic network (operated by a combination of local carriers and interexchange carriers) includes a number of end office and tandem office type central office switching systems 11. FIG. 1 shows a number of subscriber stations, depicted as telephones 1, connected to a series of central office switches 11_1 to 11_N. In the preferred implementation, the connections to the central office switches 11 utilize telephone lines, and the switches are telephone type switches for providing landline communication. However, it should be recognized that other communication links and other types of switches could be used. Trunk circuits TR carry communication traffic between the central office switches 11.

Each end office type central office switch, such as 11 and 11_N, provides switched telephone connections to and from local communication lines or other subscriber links coupled to end users stations or telephone sets 1. For example, the central office 11_1 serves as an end office to provide switched telephone connections to and from local communication lines coupled to end users telephone station sets, such as telephone 1_A, whereas the central office 11_N serves as an end office to provide switched telephone connections to and from local communication lines coupled to end users telephone station sets, such as telephone 1_B.

The typical telephone network also includes one or more tandem switching offices such as office 11_T, providing trunk connections between end offices. As such, the traffic network consists of local communication links and a series of switching offices interconnected by voice grade trunks, only two examples of which are shown as TR in FIG. 1. One set of trunks TR might interconnect the first end office 11_1 to the tandem office 11_T, whereas another set of trunks TR might interconnect the tandem office 11_T to another end office 11_N. Other trunks might directly connect end offices. Although not shown, many offices serve as both end offices and tandem offices for providing different call connections.

FIG. 1 shows connections to the stations 1 via lines, and typically these links are telephone lines (e.g. POTS or ISDN). It will be apparent to those skilled in the art, however, that these links may be other types of communication links, such as wireless links. The telephone stations may have caller ID capability. If the line is an ISDN line, the station may incorporate a display for visually presenting the caller ID information and other signaling related messages. If the link is a typical analog telephone line, the customer premises equipment includes a caller ID terminal, one example of which is shown at 5_B. The terminal 5_B displays at least telephone numbers and preferably displays alphanumeric information to enable displays of callers names.

Although shown as telephones in FIG. 1, the terminal devices or stations 1 can comprise any communication device compatible with the local communication link. However in the case of home incarceration the terminal would be a telephone terminal. The processing in accord with the instant embodiment of the invention relies on identification of the subscriber, preferably by voice based recognition. For this purpose, the terminals preferably include two-way voice communication elements.

The lines and trunks through the central offices 11 carry the communication traffic of the telephone network. The preferred telephone network, however, also includes a common channel interoffice signaling (CCIS) network carrying a variety of signaling messages, principally relating to control of processing of calls through the traffic portion of the network. The CCIS network includes packet data links (shown as dotted lines) connected to appropriately equipped central office switching systems such as offices 11 and a plurality of packet switches, termed Signaling Transfer Points (STPs) 15. To provide redundancy and thus a high degree of reliability, the STPs 15 typically are implemented as mated pairs of STPs. The CCIS network of the telephone system operates in accord with an accepted signaling protocol standard, preferably Signaling System 7 (SS7).

In the typical embodiment shown in FIG. 1, each central office 11 has at least minimal SS7 signaling capability, which is conventionally referred to as a signaling point (SP)

in reference to the SS7 network. As such, the offices can exchanges messages relating to call set-up and tear-down, typically in ISDN-UP format. At least some, and preferably all, of the central office switches 11 are programmed to recognize identified events or points in call (PICs) as advanced intelligent network (AIN) type service triggers. In response to a PIC or trigger, a central office 11 initiates a query through the CCIS signaling network to a control node to either a Service Control Point (SCP) 19 or to a database system, such as a Line Identification Database (LIDB) 21. The SCP 19 provides instructions relating to AIN type services. The LIDB 21 provides subscriber account related information, for calling card billing services or for subscriber name display purposes in an enhanced caller ID application. Those central office switching systems having full AIN trigger and query capability for communication with the SCP and/or the LIDB are referred to as Service Switching Points (SSPs).

The central office switches 11 typically consist of programmable digital switches with CCIS communications capabilities. One example of such a switch is a 5ESS type switch manufactured by AT&T; but other vendors, such as Northern Telecom and Seimens, manufacture comparable digital switches which could serve as the SSPs and SPs. The SSP type implementation of such switches differs from the SP type implementation of such switches in that the SSP switch includes additional software to recognize the full set of AIN triggers and launch appropriate queries. A specific example of an SSP capable switch is discussed in detail later, with regard to FIG. 2.

One key feature of the present invention is that the program controlled switch accepts instructions to load profiles and/or receives profiles over a signaling link. In most cases, these profiles are identified by virtual office equipment numbers. The profiles include a range of information relating to subscribers services, such as service features, classes of service and individual billing options. In accord with the preferred embodiment of the instant invention, the profiles related to persons subject to home incarceration and include features not normally found in typical telephone subscriber profiles. Thus these profiles may also include at least data relating to the identity of the individual, to facilitate caller identification which will provide the identity of the actual caller to the incarceration supervisor, agent, or controller. The profile also preferably includes call control parameters. These parameters result in limitation of outgoing calls to identified classes of called parties, and in certain instances to a limited number of specific individuals. The profile may also include a list of identified key words and telephone circuit signals for which the telephone calls will be monitored. Such words may include credit card names such as, for example, Visa, Master Card, American Express, and the like. The telephone signals may include certain combinations of DTMF signals, dial-tone, hook-flash, and other signals that indicate that proscribed action is imminent.

The above described data signalling network between the SSP type central offices 11 and the SCP 19 is preferred, but other signalling networks could be used. For example, instead of the packet switched type links through one or more STP's, a number of central office switches, an SCP and any other signaling nodes could be linked for data communication by a token ring network. Also, the SSP capability may not always be available at the local office level, and several other implementations might be used to provide the requisite SSP capability. For example, none of the end office switches may have SSP functionality. Instead, each end office would connect through a trunk to a tandem office which has the SSP capability. The SSP tandem then communicates with the SCP via an SS7 type CCIS link, as in the implementation described above. The SSP capable tandem switches are digital switches, such as the 5ESS switch from AT&T; and the non-SSP type end offices might be 1A analog type switches.

The SCP 19 may be a general purpose computer storing a database of call processing information. In the preferred implementation, the SCP 19 actually is an Integrated Service Control Point (ISCP) developed by Bell Atlantic and Bell Communications Research. The ISCP is an integrated system. Among other system components, the ISCP includes a Service Management System (SMS), a Data and Reporting System (DRS) and the actual database also referred to as a Service Control Point (SCP). In this implementation, the SCP maintains a Multi-Services Application Platform (MSAP) database which contains call processing records (CPRs) for processing of calls to and from various subscribers. The ISCP also typically includes a terminal subsystem referred to as a Service Creation Environment or SCE for programming the MSAP database in the SCP for the services subscribed to by each individual customer.

The components of the ISCP are connected by an internal, high-speed data network, such as a token ring network. The internal data network also typically connects to a number of interfaces for communication with external data systems, e.g. for provisioning and maintenance. In the preferred embodiment, one of these interfaces provides communications to and from the SCP 19 via a packet switched data network, such as the TCP/IP network 27.

The SCP may be implemented in a variety of other ways. The SCP may be a general purpose computer running a database application and may be associated with one of the switches. Another alternative is to implement a database of CPRs or the like within an STP (see e.g. Farris et al. U.S. Pat. No. 5,586,177).

The LIDB database 21 is a general purpose computer system having a signaling link interface or connection to a pair of STPs 15. The computer runs a database program to maintain a database of information relating to customer accounts and identifications. For example, a subscriber's entry in the LIDB database might include the subscriber's telephone number, a personal identification number for credit card billing purposes (in non-incarceration uses), and the subscriber's name and address.

The preferred telephone network also includes one or more intelligent peripherals (IPs) 23 to provide enhanced announcement and digit collection capabilities and speech recognition. The IP 23 is essentially similar to that disclosed in commonly assigned U.S. Pat. No. 5,572,583 to Wheeler, Jr. et al. entitled "Advanced Intelligent Network with Intelligent Peripherals Interfaced to the Integrated Services Control Point," and the disclosure of the network and operation of the IP disclosed from that Patent is incorporated herein in its entirety by reference.

The IP 23 may connect to one or more central offices 11. The connections transport both communication traffic and signaling. The connection between a central office 11 and the IP 23 may use a combination of a T1 and a Simplified Message Desk Interface (SMDI) link, but preferably this connection utilizes a primary rate interface (PRI) type ISDN link. Each such connection provides digital transport for a number of two-way voice grade type telephone communications and a channel transporting signaling data messages in both directions between the switch and the IP.

As discussed more later, there are certain circumstances in which the SCP 19 communicates with the IP 23. These communications could utilize an 1129 protocol and go through an SSP type central office 11 and the SS7 network. However, in the preferred embodiment of FIG. 1, the IP 23 and the SCP 19 communicate with each other via a separate second signaling network 27. These communications through network 27 between the IP and the SCP may utilize an 1129+ protocol or a generic data interface (GDI) protocol as discussed in the above incorporated Patent to Wheeler, Jr. et al.

The IP 23 can provide a wide range of call processing functions, such as message playback and digit collection. In the preferred system of the instant invention, the IP also performs speaker identification/verification (SIV) on audio signals received from the detainees (or subscriber's in other applications of the generic invention). Specifically, the IP 23 used for the personal dial tone service includes a voice authentication module to perform the necessary speaker identification/verification function. The IP 23 also includes storage for detainee or subscriber specific template or voice feature information, for use in identifying and authenticating detainees or subscribers based on speech.

In the simplest form, the IP 23 serving a subscriber's local area stores the templates and performs the speaker identification/verification. However, in a system serving a large geographic area and providing personal dial tone to a large, roaming subscriber base, the templates may be transferred between SCP/IP pairs, to allow an IP near a subscriber's current location to perform the speaker identification/verification on a particular call. For example, if a remote IP 23_R required a template for a subscriber from the region served by the IP 23, the remote IP 23_R would transmit a template request message through the network 27 to the IP 23. The IP 23 would transmit the requested template back through the network 27 to the remote IP 23_R.

In a network such as shown in FIG. 1, routing typically is based on dialed digit information, profile information regarding the link or station used by the calling party and profile information regarding a line or station in some way associated with the dialed digits. Each exchange is identified by one or more three digit codes. Each such code corresponds to the NXX digits of an NXX-XXXX (seven digit) telephone number or the three digits following the area code digits (NPA) in a ten-digit telephone number. The telephone company also assigns a telephone number to each subscriber line connected to each switch. The assigned telephone number includes the area code and exchange code for the serving central office and four unique digits.

Central office switches utilize office equipment (OE) numbers to identify specific equipment such as physical links or circuit connections. For example, a subscriber's line might terminate on a pair of terminals on the main distribution frame of a switch 11. The switch identifies the terminals, and therefore the particular line, by an OE number assigned to that terminal pair. For a variety of reasons, the operating company may assign different telephone numbers to the one line at the same or different times. For example, a local carrier may change the telephone number because a subscriber sells a house and a new subscriber moves in and receives a new number. However, the OE number for the terminals and thus the line itself remains the same.

On a normal call, an end office type switch will detect an off-hook condition on the line and provide dial tone. The switch identifies the line by its OE number. The office also retrieves profile information corresponding to the OE number and off-hook line. If needed, the profile identifies the currently assigned telephone number. The switch in the end office receives dialed digits and routes the call. The switch may route the call to another line serviced by that switch, or the switch may route the call over trunks and possibly through one or more tandem offices to an office that serves the called party's station or line. The switch terminating a call to a destination will also utilize profile information relating to the destination, for example to forward the call if appropriate, to apply distinctive ringing, etc.

AIN call processing involves a query and response procedure between an SSP capable switching office 11 and a database system, such as the SCP 19. The SSP capable switching offices initiate such processing upon detection of triggering events. At some point during processing of a telephone call, a central office switching system 11 will recognize an event in call processing as a 'Point in Call' (PIC) which triggers a query to the SCP 19. Ultimately, the SCP 19 will return an instruction to the switching system 11 to continue call processing. This type of AIN call processing can utilize a variety of different types of triggers to cause the SSPs 11 to initiate the query and response signaling procedures with the SCP 19. In the example discussed below, the personal dial tone service utilizes an off-hook immediate trigger, a dialed number trigger and a terminating attempt trigger (TAT), to facilitate different aspects of that service.

In accord with one aspect of the present invention, before providing dial-tone service, the SSP central office 11 that is serving an outgoing call extends the call to the IP 23 providing the speaker identification/verification (SIV) functionality. In the preferred embodiments, this operation involves AIN type call routing to the IP. The IP 23 prompts the caller and collects identifying information, preferably in the form of speech. The IP analyzes the caller's input to identify the caller as a particular subscriber or detainee. If successful, the IP signals the SSP to load profile data for that subscriber or detainee into the register assigned to the call in the call store. In most of the preferred service applications, the IP disconnects, and the SSP central office 11 processes the call in accord with the loaded profile information. For example, the central office 11 may now provide actual dial tone or provide a message prompting the caller to dial a destination number. The caller dials digits, and the central office processes the digits to provide the desired outgoing call service, in the normal manner. The IP may stay on the line, to monitor speech and thus caller identity, and to monitor for prestored key words or telephone signals for the home incarceration application which constitutes the preferred embodiment of the present invention.

The call processing by the central office switch 11 utilizes the loaded subscriber or detainee profile information. For example, the profile data may indicate specific procedures for billing the call to this subscriber on some account not specifically linked to the originating telephone station or line. For example, in a college dormitory, the billing information might specify billing of a student's calls to the account of the student's parent(s). Any call restrictions, imposed at the wish of the parents, would be reflected in the profile. The switch would restrict the calling services accordingly, e.g. to limit distance, cumulative cost and/or duration of calls. In the preferred home incarceration embodiment the profile reflects restrictions and conditions associated with the individual detainees.

The inventors also envision use of selected subscriber profile information on incoming calls. When a serving central office SSP 11 detects a call to a line having the service of the invention, processing hits a terminating

attempt trigger (TAT). The SSP interacts with the SCP 19 and routes the call to the IP 23. The IP 23 prompts the caller to identify a desired called party, e.g. one of the detainees sharing the telephone instrument or terminal. Menu announcement together with either digit collection or preferably speech recognition processing by the IP 23 facilitates identification of the desired called party from those associated with the line. Based on identification of the called party, the IP 23 signals the SSP switch 11 to load profile data for that party into the register assigned to the call in the call store. In this case, however, the switch 11 uses selectively loaded profile information for terminating the call. The IP disconnects, and the SSP central office 11 processes the call in accord with the loaded profile information.

For example, the central office 11 may provide a distinctive ringing signal corresponding to the identified called party. This service enables distinctive ringing for multiple parties on one line without assigning each party a separate telephone number. In non-incarceration embodiments the loaded profile information may specify call forwarding in event of a busy or no-answer condition. This enables routing of the call to the identified subscriber's mailbox, or another alternate destination selected by the subscriber, even though the call did not utilize a unique telephone number uniquely assigned to the called subscriber.

The present invention also encompasses a procedure in which a subscriber calls in from a line not specifically designated for personal dial tone service. In the incarceration embodiment this may occur in the case of detainees who are allowed pre-determined release time. The network routes the call to the IP 23, and the IP identifies the caller and the line from which the call was made. The caller can interact with the IP 23 to have her personal dial tone service associated with that line, either for one call or for some selected period of time. The IP 23 instructs the appropriate central office switch(es) 11 to load profile data associated with the caller.

The IP 23 might instruct the end office switch to load the profile data only in the assigned call store register. The switch would use the profile data only for a single call, for example to bill a call from a pay-phone or a hotel room telephone to the subscriber's home account in the situation where the detainee is allowed this privilege. Alternatively, the IP 23 might instruct the central office 11 serving the line to the calling station 1 to utilize a virtual office equipment number (OE) and associated profile data for calls to and from that line for some period of time. In this later example, the IP 23 would also instruct the central office 11 serving the line to the subscriber's home station 1 to modify the subscriber's profile to forward calls for the subscriber's telephone number. The modified profile data in the home office 11 would result in forwarding of the subscriber's incoming calls through the office 11 to the selected station 1, for the set period of time.

The present invention relies on the programmable functionality of the central office switches and the enhanced call processing functionalities offered by the IPs. To understand these various functionalities, it may be helpful to review the structure and operation of a program controlled central office and one implementation of an IP. Subsequent description will explain several of the above outlined call processing examples in greater detail.

FIG. 2 is a simplified block diagram of an electronic program controlled switch which may be used as any one of the SSP type central offices 11 in the system of FIG. 1. As illustrated, the switch includes a number of different types of modules. In particular, the illustrated switch includes interface modules 51 (only two of which are shown), a communications module 53 and an administrative module 55.

The interface modules 51 each include a number of interface units 0 to n. The interface units terminate lines from subscribers' stations, trunks, T1 carrier facilities, etc. Each such termination is identified by an OE number. Where the interfaced circuit is analog, for example a subscriber loop, the interface unit will provide analog to digital conversion and digital to analog conversion. Alternatively, the lines or trunks may use digital protocols such as T1 or ISDN. Each interface module 51 also includes a digital service unit (not shown) which is used to generate call progress tones and receive and detect dialed digits in pulse code or dual-tone multifrequency form.

In the illustrated embodiment, the unit 0 of the interface module 51' provides an interface for the signaling and communication links to the IP 23. In this implementation, the links preferably consist of one or more ISDN PRI circuits each of which carries 23 bearer (B) channels for communication traffic and one data (D) channel for signaling data.

Each interface module 51 includes, in addition to the noted interface units, a duplex microprocessor based module controller and a duplex time slot interchange, referred to as a TSI in the drawing. Digital words representative of voice information are transferred in two directions between interface units via the time slot interchange (intramodule call connections) or transmitted in two directions through the network control and timing links to the time multiplexed switch 57 and thence to another interface module (intermodule call connection).

The communication module 53 includes the time multiplexed switch 57 and a message switch 59. The time multiplexed switch 57 provides time division transfer of digital voice data packets between voice channels of the interface modules 51 and transfers signaling data messages between the interface modules. The switch 57 together with the TSIs of the interface modules form the overall switch fabric for selectively connecting the interface units in call connections.

The message switch 59 interfaces the administrative module 55 to the time multiplexed switch 57, so as to provide a route through the time multiplexed switch permitting two-way transfer of control related messages between the interface modules 51 and the administrative module 55. In addition, the message switch 59 terminates special data links, for example a link for receiving a synchronization carrier used to maintain digital synchronism.

The administrative module 55 provides high level control of all call processing operations of the switch 11. The administrative module 55 includes an administrative module processor 61, which is a computer equipped with disc storage 63, for overall control of CO operations. The administrative module processor 61 communicates with the interface modules 51 through the communication module 55. The administrative module 55 may include one or more input/output processors (not shown) providing interfaces to terminal devices for technicians and data links to operations systems for traffic, billing, maintenance data, etc.

A CCIS terminal 73 and an associated data unit 71 provide an SS7 signalling link between the administrative module processor 61 and one of the STPs 15 (see FIG. 1). Although only one such link is shown, preferably there are a plurality of such links providing redundant connections to both STPs of a mated pair and providing sufficient capacity to carry all necessary signaling to and from the particular office 11. The

SS7 signaling through the terminal 73, the data unit 71 and the STPs provides two-way signaling data transport for call set-up related messages to and from other offices. These call set-up related messages typically utilize the ISDN-UP (ISDN-users part) protocol portion of SS7. The SS7 signaling through the terminal 73, the data unit 71 and the STPs also provides two-way signaling data transport for communications between the office 11 and database systems or the like, such as the SCP 19. The communications between the office 11 and the database systems or the like utilize the TCAP (transactions capabilities applications part) protocol portion of SS7.

As illustrated in FIG. 2, the administrative module 55 also includes a call store 67 and a program store 69. Although shown as separate elements for convenience, these are typically implemented as memory elements within the computer serving as the administrative module processor 61. The program store 69 stores program instructions which direct operations of the computer serving as the administrative module processor 61.

For each call in progress, a register assigned within the call store 67 stores translation and user profile information retrieved from disc storage 63 together with routing information and any temporary information needed for processing the call. For example, for a residential customer initiating a call, the call store 67 would receive and store line identification and outgoing call billing information corresponding to an off-hook line initiating a call. For the personal dial-tone service, the assigned register in the call store 67 will receive and store different profile data depending on the particular subscriber or detainee associated with any given call. A register in the call store is assigned and receives profile data from the disc memory both for originating subscribers or detainees on outgoing calls and for terminating subscribers or detainees on incoming calls.

A variety of adjunct processor systems known in the telephone industry can be used as the IP 23. The critical requirements are that the IP system process multiple calls and perform the subscriber identification functions, preferably by speaker identification and authentication. FIG. 3 is a functional diagram illustration of an IP 23 for performing the subscriber identification functions, possibly by dialed digit input and preferably by analysis and recognition of speech.

The preferred IP architecture utilizes separate modules for different types of services or functions, for example, one or two Direct Talk type voice server modules 231A, 231B for interfacing ISDN PRI trunks to the SSP central office(s) 11. Separate modules 233, 235 perform voice authentication and speech recognition. The IP 23 includes a variety of additional modules for specific types of services, such as a server module 237 for fax mail, and another server 239 for voice mail services. The various modules communicate with one another via an internal data communication system or bus 240, which may be an Ethernet type local area network.

Each Direct Talk module 231A or 231B comprises a general purpose computer, such as an IBM RS-6000, having digital voice processing cards for sending and receiving speech and other audio frequency signals, such as IBM D-talk 600 cards. Each voice processing card connects to a voice server card which provides the actual interface to T1 or primary rate interface ISDN trunks to the switching office. In the PRI implementation, the Direct Talk computer also includes a signaling card, providing two-way signaling communication over the D-channel of the PRI link. Each Direct Talk computer also includes an interface card for providing two-way communications over the internal data communications system 240.

The voice processing cards in the Direct Talk modules 231A, 231B provide voice message transmission and dialed digit collection capabilities. The modules 231A, 231B also perform the necessary line interface functions for communications to and from those servers which do not incorporate actual line interfaces. For example, for facsimile mail, a Direct Talk module 231 connected to a call would demodulate incoming data and convert the data to a digital format compatible with the internal data communication network 240. The data would then be transferred over network 240 to the fax server 237. For outgoing facsimile transmission, the server 237 would transfer the data to one of the Direct Talk modules over the network 240. The Direct Talk module 231 would reformat and/or modulate the data as appropriate for transmission over the ISDN link to the switch 11.

The Direct Talk modules provide a similar interface function for the other servers, such as the voice mail server 239, the speech recognition module 235 and the voice authentication module 233. For incoming speech signals, the Direct Talk module connected to a call receives digital speech signals in the standard pulse code modulation format carried on a B-channel of an ISDN link. The Direct Talk module reformats the speech data and transmits that data over the internal network 240 to the server or module performing the appropriate function, for example to the authentication module 233 for analysis and comparison of features to stored templates or feature data for known subscribers.

In the outgoing direction, the currently connected Direct Talk module may play an announcement from memory, e.g. to prompt a caller to say their name. Alternatively, the Direct Talk module may receive digitized speech over the network 240 from one of the other modules, such as a stored message retrieved from voice mail server 239. The Direct Talk module reformats the speech signal as needed for transmission over the ISDN B-channel to the caller.

The illustrated IP also includes a communication server 243. The communication server 243 connects between the data communication system 240 and a router 241, which provides communications access to the TCP/IP network 27 that serves as the second signaling communication system. The communication server 243 controls communications between the modules within the IP 23 and the second signaling communication system. The server 243 and the router 241 facilitate communication between the elements of the IP 23 and the SCP 19. The IP may also use this communication system to communicate with other IP's, for example to send subscriber voice template information to the remote IP 23_R (FIG. 1) or to receive such information from that IP or some other network node.

The personal dial tone service relies on the voice authentication module 233 to perform the necessary speaker identification/verification function. For the identification and authentication of subscribers, the voice authentication module 233 within the IP 23 stores a template or other feature or voice pattern information for each person who has the personal dial tone service in the area that the IP services. For example, if the subscriber utilizes the personal dial tone service from a particular line, such as a shared line in an incarceration facility or a dormitory or the like, the IP stores the subscriber's voice pattern information in a file associated with the office equipment (OE) number of the particular line. If the IP 23 serving a call does not store the template or feature data for a particular subscriber, the IP 23 may obtain

subscriber identification by dialed digit input and then obtain a copy of the template or feature data from a remote IP 23_R via communication through the TCP/IP network 27, in order to authenticate the subscriber's identity.

Using current technology, a new subscriber or detainee would get on line with the IP serving that subscriber and 'train' that IP by speaking certain words and phrases. From the received audio signals representing those words and phrases, the IP would store templates or other pattern information for use in identifying and/or verifying that a caller is the particular subscriber or detainee.

During actual call processing, the voice authentication module 233 receives speech information from the caller. The voice authentication module 233 compares the received information to its stored template or feature data to identify a calling party as a particular subscriber or detainee.

The present invention also relies on the speech recognition capability of the module 235, particularly in processing of incoming calls in certain situations. The speech recognition module 235 enables the IP to analyze incoming audio information to recognize vocabulary words. The IP 23 interprets the spoken words and phrases to determine subsequent action. For example, the IP might recognize the caller speaking the name of a called subscriber or detainee and use the subscriber or detainee identification to instruct the terminating central office to control the call in accord with that subscriber or detainee's profile.

The preferred routing of the calls in accord with the invention utilizes AIN type call processing. To understand the call processing, it may be helpful to consider several specific examples in more detail.

In a first example, consider an outgoing call from the station 1_A to the station 1_B. Assume per call assignment of profile data to the originating line, for personal dial tone service on each outgoing call. FIG. 4 provides a simplified flow diagram of the signal flow and processing for such an outgoing call.

Assume use of a standard telephone for purposes of this example. The person lifts the handset creating an off-hook state in the telephone 1_A, and a corresponding signal or change in state on the line to the central office 11 (step S1). In this call flow, the off-hook signal is a type of service request, i.e. a request to make an outgoing call. The serving central office 11_1 detects the off-hook and commences its call processing. Specifically, the central office assigns a register in the call store 67 to this call and loads profile information associated with the off-hook line from the disc storage 63 into the assigned register. In this case, the central office 11_1 is an SSP capable office, and the loaded profile data indicates an off-hook immediate trigger set against the particular line. The serving SSP type office 11_1 therefore detects this off-hook PIC as an AIN trigger (step S2).

In response to the off-hook and the off-hook trigger set in the subscriber's profile, the SSP type central office switch 11_1 launches a query to the SCP 19 (step S3). Specifically, the SSP 11_1 creates a TCAP query message containing relevant information, such as the office equipment (OE) number assigned to the off-hook line, and transmits that query over an SS7 link to one of the STPs 15. The query includes a destination point code and/or a global title translation addressing the message to the SCP 19, and the STP 15 relays the query message over the appropriate link to the SCP 19. The query from the SSP central office 11_1 identifies the caller's line by its associated office equipment (OE) number and possibly by a single telephone number associated with the off-hook line.

In response to a query, the SCP 19 accesses its a database, typically, the MSAP database set up in the ISCP, to determine how to process the particular call. The SCP 19 identifies an access key in the query and uses the key to retrieve the appropriate record from the database. In this case, the query indicates an off-hook trigger as the trigger event, therefore the SCP 19 uses the calling party office equipment (OE) number as the access key. The SCP 19 retrieves a call processing record (CPR) corresponding to the office equipment (OE) number associated with the off-hook line and proceeds in accord with that CPR (step S4).

For the present example of the personal dial tone service, the CPR will provide information necessary for routing the call to some node of the network that will perform speaker identification/verification (SIV). In the preferred embodiment, the SIV is a function performed by an Intelligent Peripheral (IP), therefore the CPR provides information for routing the call to the nearest available IP having the SIV capability.

Based on the CPR, the SCP 19 formulates a response message instructing the SSP central office 11_1 serving the customer to route the call. In this case, the message includes information, e.g. a office equipment (OE) number or telephone number, used for routing a call to the identified IP 23. The SCP 19 formulates a TCAP message in SS7 format, with the destination point code identifying the SSP office 11_1. The SCP 19 transmits the TCAP response message back over the SS7 link to the STP 15, and the STP 15 in turn routes the TCAP message to the SSP central office 11_1 (see step S5).

The SSP type switch in the central office 11_1 uses the routing information to connect the call to one of the lines or channels to the IP 23. A two-way voice grade call connection now extends between the calling station 1_A and the IP 23 (step S6). In the present example, the switch actually connects the off-hook line to the line to the IP before providing dial tone.

As noted above, the communication link to the IP 23 provides both line connections and signaling, preferably over a primary rate interface (PRI) type ISDN link. When the central office 11_1 extends the call from the calling party's line to a line circuit (over a B channel) to the IP 23, the switch in that office also provides call related data over the signaling link (D channel for ISDN). The call related data, for example, includes the office equipment (OE) number normally associated with the off-hook line and possibly the telephone number for that line.

In response to the incoming call, the IP 23 will seize the line, and it will launch its own query to the SCP 19 (step S7). In the preferred network illustrated in FIG. 1, the IP 23 and the SCP 19 communicate with each other via a separate second signalling network 27, for example utilizing either an 1129+ protocol or a generic data interface (GDI) protocol as discussed in U.S. Pat. No. 5,572,583 to Wheeler, Jr. et al. The query from the IP 23 again identifies the caller's line by at least its associated office equipment (OE) number.

In response to the query from the IP 23, the SCP 19 again accesses the appropriate CPR (step S8) and provides a responsive instruction back through the network 27 to the IP 23 (step S9). Although the IP 23 could passively monitor any speech that the user might utter, the preferred implementation utilizes a 'Challenge Phase' to prompt the user to input specific identifying information. In this case, the instruction causes the IP 23 to provide a prompt message over the connection to the caller (step S10). Here, the signal to the caller may be a standard dial tone or any other appropriate audio signal. Preferably, the instruction from the SCP 19

causes the IP 23 to provide an audio announcement prompting the caller to speak personal information. In one preferred example, in step S10 the IP plays an audio prompt message asking the caller, 'Please say your full name'. The process may ask for any appropriate identifying information.

The signal received by the IP 23 goes over the lines and through the central office switch(es) for presentation via the off-hook telephone 1_A to the calling party. In response, the caller will speak identifying information into their off-hook telephone, and the network will transport the audio signal to the IP 23 (step S11).

As noted above, an IP 23 can provide a wide range of call processing functions, such as message playback and digit collection. In the preferred system, the IP also performs speaker identification/verification (SIV) on the audio signal received from the off-hook telephone in step S11. When the IP 23 receives speech input information during actual call processing, for this service example, the IP analyzes the speech to extract certain characteristic information (step S12).

The IP 23 stores a template or other voice pattern information for each person who has the personal dial tone service in the area that the IP normally services. If the IP 23 does not store the particular template or feature information it needs to process a call, the IP 23 can communicate with a remote IP 23_R to obtain that information. In the present shared line example, the IP 23 will store template or feature data for each subscriber or detainee associated with the particular off-hook line.

When the IP 23 receives input speech and extracts the characteristic information during actual call processing, the IP compares the extracted speech information to stored pattern information, to identify and authenticate the particular caller. In the present example, the voice authentication module 233 in the IP 23 compares the extracted speech information to the stored template or feature data for each subscriber associated with the particular off-hook line.

In step S13, the IP 23 determines if the information extracted from the speech input matches any of the stored template data feature data for an identifiable subscriber or detainee (within some threshold level of certainty). If there is a match, the IP now knows the identity of the calling subscriber or detainee. Based on the identification of the calling subscriber or detainee, the IP 23 selects a virtual office equipment (OE) number from storage that corresponds to the subscriber or detainee.

The IP 23 formulates a D-channel signaling message containing the virtual office equipment (OE) number together with an instruction to load that OE number into the register assigned to the call in place of the OE number of the off-hook line. The IP 23 supplies the message to the SSP central office switch 11, over the D-channel of the ISDN PRI link (step S14). In response, the administrative module processor 61 rewrites the OE number in the register assigned to the call using the OE number received from the IP 23.

Upon rewriting the OE number in the register, the administrative module processor 61 of central office switch 11, also reloads the profile information in the register (step S15). Specifically, the administrative module processor 61 retrieves profile information associated with the virtual office equipment (OE) number from the disc storage 63 into the register. As such, the profile information in the assigned register in the call store 67 now corresponds to the identified subscriber or detainee, rather than to the off-hook line.

The profile information provides a wide range of data relating to the subscriber or detainee's services. The profile data provides necessary billing information, enabling billing from the call to this particular subscriber or detainee for a personal call and/or billing to the incarceration supervision entity. The profile also defines various service features and restrictions applicable to this subscriber or detainee on outgoing calls, such as three-way calling in the case of an allowed service, and prohibited and/or allowable calls in the case of a detainee. The profile may also include a list of key words for which monitoring is maintained during calls. The profile may define a class of calling service available to the subscriber or detainee. In a dormitory example, the caller may be allowed a set dollar amount for long distance calls per month (e.g. $50.00). The profile data will indicate the remaining amount at the time of the call and will cause the switch to interrupt service when the available amount is exhausted. Long distance calls are usually proscribed in the case of detainees. To this end the key word list may contain credit card names and DTMF sequences which precede long distance calls. Other class of service restrictions might enable long distance calls only if collect and/or only if calling one or two specified numbers (e.g. only to the parents' house) in the dormitory situation. The class of service might enable only long distance calls within a region or country but not international calls.

In the presently preferred implementation, when the central office switch 11, reloads the profile, the central office disconnects the link to the IP 23 and connects tone receivers to the caller's line. Optionally, the central office 11, may provide a 'dial tone' or other message over the line (step S16). The caller now dials digits in the normal manner (step S17), and the switch in the central office 11, loads the dialed digits into the assigned register within the call store 67. The central office 11, utilizes the dialed digits and the subscriber's profile data to process the call (S18). For example, if the dialed digits represent a call within the subscriber's permitted class of service, the switch completes the call to the destination station 1_B using the dialed digits in the normal manner. If the profile data requires a particular billing treatment, e.g. to bill a long distance call to the subscriber, the switch makes the appropriate record and forwards the record to the exchange carrier company's accounting office equipment. In accord with another aspect of the invention, the network provides caller ID data naming the identified subscriber or detainee to the destination station.

The processing to complete the call, performed in step S18, actually involves a sequence of steps. Of particular note, some of these steps facilitate delivery of caller ID information to the destination station. The present invention involves delivering caller ID information which corresponds to the identified subscriber or detainee, preferably the subscriber or detainee's name, rather than simply the number of the line or station from which the subscriber or detainee initiates the call. Two processing methodologies are envisioned for providing this calling subscriber or detainee ID feature, one involving access to name information in a central database such as LIDB and the other relying on name data from the subscriber or detainee's profile.

FIG. 4B is a simplified process and signal flow diagram, illustrating the call completion operations, including caller ID display using data from the profile. The network performs the steps depicted in FIG. 4B after identification of the subscriber or detainee, preferably based on speaker identification/verification (SIV). As discussed earlier, the IP 23 supplies the signaling message containing the virtual office equipment (OE) number and the instruction to load that OE number into the assigned register to the SSP central office switch 11, over the D-channel of the ISDN PRI link

(step S14). In response, the administrative module processor 61 rewrites the OE number in the register and reloads the profile information in the register (step S15).

The central office 11_1 provides dial tone or the like over the line (step S16), the caller dials digits corresponding to the desired destination (step S17), and the switch in the central office 11_1 begins its processing to route the call through the network. Initially, the central office 11_1 uses the dialed number to initiate a CCIS communication with the exchange serving the intended destination, in the example the terminating central office 11_N.

Specifically, the subscriber or detainee's serving central office 11_1 generates an Initial Address Message (IAM) for transmission to the terminating central office 11/ (S181). The IAM message includes the SS7 destination point code (DPC) of the terminating central office 11_N and the SS7 origination point code (OPC) of the customer's serving-end central office 11_1, for addressing purposes. The payload portion of the IAM message includes the called and calling numbers. In accord with the invention, the originating end office 11_1 reads name data from the identified subscriber's profile, currently loaded in the assigned register, and places that data in additional field of the IAM message or in an accompanying information message addressed in the same manner as the IAM message. The originating central office transmits the IAM message and possibly an accompanying information message through the CCIS network to the distant terminating office 11_N (step S181).

When the terminating office 11_N receives the IAM message, the administrative module processor for that office retrieves the customer profile for the number in the destination number field of that message (e.g. the number for the telephone 1_B) from its mass storage system and loads that profile into one of its call store registers. If the called party has an enhanced caller ID service, with name display, the terminating central office 11_N would normally recognize the attempt to complete to that party's number message as a terminating attempt trigger (TAT) type point in call (PIC) to trigger access to the LIDB database for name information. However, in this embodiment of the invention, the terminating end office detects the receipt of the subscriber's name data with the IAM message, therefore the administrative module processor in that office overrides the trigger.

The terminating central office switching system 11_N transmits an Address Complete Message (ACM) back to the central office 11_1 and if the called line is available applies ringing signal to the called party's line (S182). The ACM includes a variety of information, including a calling party status indicator, e.g. line free or busy. If the line is not busy, the end office 13 rings the station Y corresponding to the dialed digits 703-333-5678, and generates the appropriate indicator in the Address Complete Message (ACM) to indicate that it received the request for a call and that the number is not busy. The ACM message is sent back by simply reversing the point codes from the IAM message. Now the destination point code (DPC) is the point code of the central office 11_1 and the origination point code (OPC) is the point code of the central office 13. In response to the ACM message, if the called line is available, the originating central office 11 applies a ringback tone signal to the line to the calling station 1_A (S183).

As part of its operations to ring the called telephone station, the terminating central office 11_N transmits a caller ID signal over the line. If the called party has ISDN service or the like, the switch sends a signaling message along with the ringing signal. If the called party has analog telephone service, the switch 11_N transmits a caller ID message (step S184) as frequency shift keyed (FSK) data inserted in the silent interval between the first ringing signal (step S182) and the second ringing signal (S185) applied to the called party's line.

In accord with the invention, the caller ID message applied to the called party's line includes the telephone number associated with the calling station 1_A and at least some additional data specific to the identified subscriber. If the called party has enhanced caller ID for displaying name data, the ISDN telephone or the caller ID terminal 5_B receives the number and the name data received with the IAM message in step S181. The caller ID terminal 5_B or a display device in the ISDN telephone displays the received number and name information, identifying the actual calling party, for review before the called party chooses to answer the call.

If the called party subscribes only to normal caller ID, the end office switch 11_N can transmit only a limited amount of information. For this purpose, the switch will select and transmit one or two characters from the subscriber identification data along with the telephone number. For example, if four persons normally call from the particular originating telephone station or line, the data sent to the terminating central office 11_N might include a letter or number identifying each subscriber. The switch 11_N would transmit that letter or number with the telephone number in the caller ID message for display.

If someone answers the telephone station 1_B, the terminating central office switching system 11_N detects an off-hook condition (S13) and sends an Answer Message (ANM) back to the originating central office 11_1 through one or more of the STPs 15. The ANM message indicates that the called telephone 1_B was picked up. Also, at that time the actual telephone traffic trunk circuit is connected together between the central offices 11_1 and 11_N The central offices 11 connect the lines to the stations to the respective ends of the trunk circuit, to complete the voice path. At this point, actual voice communication is established between the calling station 1_A and the called station 1_B. Communication continues until one or both parties hang up, at which time, all of the switched connections are torn down.

FIG. 4C is a simplified process and signal flow diagram, illustrating the call completion operations including, caller ID display involving access to name information in a central LIDB database. The network performs the steps depicted in FIG. 4C after identification of the subscriber, preferably based on speaker identification/verification (SIV). As in the example of FIG. 4B, the central office switch 11_1 receives an instruction containing the subscriber's virtual office equipment (OE) number (step S14), loads the corresponding profile information in the register (step S15) and sends dial tone or the like over the line (step S16). The subscriber dials digits corresponding to the desired destination (step S17), and the switch in the central office 11_1 transmits an IAM message through the interoffice signaling network to the terminating central office 11_N. The information sent in or with the IAM message in step S191, however, is different than in the earlier example.

In this embodiment, the originating end office 11_1 reads a short code identifier from the identified subscriber's profile, currently loaded in the assigned register, and places that identifier in additional field of the IAM message or in an accompanying information message addressed in the same manner as the IAM message. For example, if the network provides personal dial tone service to four identified persons

associated with the originating telephone 11_1, the short code might comprise a number from zero to three or letters such as A, B, C and D, identified by the state of two bits in the IAM or accompanying information message.

As in the earlier example, the originating end office 11_1 addresses and transmits the IAM message with the specific subscriber identifier code through the SS7 signaling network for receipt by the terminating office 11_N. If the called party has only normal caller ID service, then the terminating office 11_N would transmit a normal caller ID message to the destination, with the identifier appended to the calling party telephone number as an extra digit or character. If the called party often receives calls from this subscriber, even the limited subscriber specific identification provided by the code will enable the called party to recognize that the current call is from the identified subscriber.

FIG. 4B depicts the processing steps, beginning in step S192, for processing a call to a called customer having the enhanced caller ID service for name and number display. In such a case, when the terminating office 11_N, the administrative module processor in that office loads the profile for the called subscriber's telephone number into a register in the call store assigned to this call. Of particular note, because the called customer has the enhanced name and number type caller ID service, the customer profile record establishes a terminating attempt trigger (TAT) against the that customer's telephone number.

At this point, the terminating office 11_N recognizes the called party telephone number in the destination number field of the IAM message as a terminating attempt trigger (TAT) type point in call or PIC (step S192). In response to this PIC, the terminating office 11_N launches a second query message through one or more of the STP(s) 15 to the LIDB database 21 (step S193). The query message includes both the telephone number associated with the calling station 1_A or its telephone line as well as the code identifying the specific subscriber or detainee making that call.

The LIDB database 21 uses the calling party telephone number and the code identifying the specific subscriber or detainee, received in the query, to retrieve that one subscriber or detainee's account file record from the database (step S194). The query also indicates the cause of the query, i.e. the TAT triggering event. From this information, the LIDB database recognizes that the query is a request for name information. The database 21 therefore reads up to 15 characters of name data from the subscriber or detainee's account file. The LIDB database 21 compiles a TCAP call control message including the name data and returns that call control message to the terminating central office 11_N via the SS7 network.

The terminating central office switching system 11_N receives the call control message from the LIDB database 21.

To provide the caller ID service in this embodiment, the terminating end office 11_N combines the name data from the call control message together with the calling party number as two caller ID messages. The end office 11_N then signals the originating office 11_1 and initiates ringing of the called party's line, as discussed in more detail below.

Assuming for this discussion that the called line is available, the terminating central office switching system 11_N transmits an Address Complete Message (ACM) indicating availability back to the central office 11_1 and applies ringing signal to the called party's line (step S182). In response to the ACM message, if the called line is available, the originating central office 11 applies a ringback tone signal to the line to the calling station 1_A (S183)

As part of its operations to ring the called telephone station, the terminating central office 11_N transmits a caller ID signal over the line. If the called party has ISDN service or the like, the switch sends the caller ID signaling messages along with the ringing signal. If the called party has analog telephone service, the switch 11_N transmits the caller ID messages sequentially over the line (step S184) as frequency shift keyed (FSK) data inserted in the silent interval between the first ringing signal (step S182) and the second ringing signal (S185) applied to the called party's line. As in the earlier example, the display provides the telephone number associated with the calling station 1_A as well as the name data for the specifically identified calling subscriber or detainee.

In the shared line example, each person normally expected to use the line to station 1_A is a different subscriber to the personal dial tone service. As the subscribers or detainees make outgoing calls, they each receive their own individualized service over the line on each separate call, in precisely the manner described above relative to steps S1 to S18 and the personal caller ID as described above relative to FIGS. 4B and 4C. For example, each subscriber or detainee may receive a different level of calling privileges and/or class of service based on their ability and/or desire to pay for telephone services. Likewise each detainee may receive a different level of restriction based upon court determined conditions. Also, the called party receives caller ID information including both the origination telephone number and the name or other identifying information associated specifically with the calling subscriber or detainee.

Returning to step S13 in FIG. 4A, the extracted information characterizing the input speech signals may not match any of the templates or feature data used by the IP 23. In this event, the process flows to step S19. The IP will count the number of tries or attempts to identify the subscriber and permit some maximum number of failed attempts (N). Assume, for example, that the software allows only two identification attempts on one call (N=2). On the first failure, the number of tries is less than N, therefore processing returns to step S10, and the IP 23 again transmits the prompt for speech input. The caller again speaks the requested input information (S11), and the authentication module 233 again analyzes the input information (S12). If the second input adequately matches a stored subscriber's information in step S13, the processing flows through steps S14 to S18 to complete the call as described above.

However, if the extracted speech information does not match a stored subscriber template or feature data, processing again flows to step S19. If the number of tries now corresponds to the limit N, for example on the second failed attempt, the processing branches to step S20. The IP 23 may now transmit a message indicating denial of service, although this is optional. In the case of detainees the IP 23 also transmits a message indicating failure of identification and denial of service to the supervising authority central data node.

The IP 23 formulates a D-channel signaling message instructing the central office switch 11_1 to process the call in accord with default conditions and transmits that instruction to the central office switch (step S21). The instruction could include a default OE number corresponding to a default profile, or the message could instruct the switch to proceed using the OE and profile data for the off-hook line itself. The IP 23 supplies the message to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link (step S21). The administrative module processor 61 resumes call processing using the appropriate default OE and profile. In the case of

detainees, the default profile may set up a call to a live attendant in order to obtain additional information which may indicate a problem in incarceration conditions. In all identification failures involving detainees records are maintained in the supervisory data node and an alarm condition may be signaled.

In the preferred embodiment, the switch provides a normal dial tone (S22), collects dialed digits from the caller (S23) and processes the call (S24). However, the default profile provides only some limited class of service, for example only emergency 911 service. The default call processing provides no additional information from the profile corresponding to any particular subscriber or detainee, therefore the network processes the call as a normal call for caller ID purposes. The caller ID service will provide only the telephone number to callers having normal caller ID, and the network will access LIDB database 21 to provide name information if any name is associated strictly with the telephone number, essentially in the manner that the network provides such services when there is no personalized dial tone service involved. In the detainee situation the caller ID signaling may include an alarm indication.

In the above example, the network disconnected the IP 23 after identifying the subscriber or detainee and providing the subscriber or detainee's virtual OE number to the serving central office 11. For some applications of the personal dial tone service, particularly in the case of home incarceration, the central office 11 would maintain a bridged connection of the IP 23 on the line, to enable the IP to monitor the call. For example, in a detainee telephone service, each detainee typically would have only limited telephone rights as specified in each detainee's profile data. To prevent one detainee from selling their telephone service rights to another detainee, the IP 23 would periodically or constantly monitor the outgoing speech signals from the incarceration facility line.

The voice authentication module 233 would initially identify the detainee subscriber as discussed above, and would periodically recheck to authenticate the identity of the party using the line. If the voice authentication module detects some other party using the line or did not detect the identified subscriber or detainee's speech for some predefined time interval, the IP 23 would instruct the serving central office switch 11 to disconnect the call. The IP 23 may send messages to the switch or to other network elements to initiate additional action, such as profile modification to further limit a particular detainee's telephone privileges and/or to notify incarceration authorities of misuse of telephone privileges.

The first detailed example discussed above related to personal dial tone service provided on a per-call basis on a shared use line. Several known subscribers or detainees may routinely use their personal dial tone service over the same line. As noted earlier, an alternate form of the personal dial tone service can be activated on a dial-up basis. Consider now an example of a dial-up activation for a single call.

For this example, assume that a subscriber or detainee's normal or 'home' telephone is telephone 1_B. The end office switch 11_N stores the subscriber or detainee profile data for the line associated with that telephone station. Now assume that the subscriber or detainee is using station 1_A connected through a telephone line to central office 11_1. FIG. 5 provides a simplified flow diagram of the signal flow and processing for such a call.

The subscriber lifts the handset creating an off-hook state in the telephone 1_A and a signal to office 11 (step S31). The serving central office 11_1 detects the off-hook and commences its call processing. Specifically, the central office assigns a register in the call store 67 to this call and loads profile information associated with the off-hook line from the disc storage 63 into the register. In this case, the profile data associated with the line does not provide an off-hook trigger because the line is not specifically associated with the shared use type personal dial tone service discussed above. The central office 11_1 therefore provides dial tone in the normal manner (step S32).

If making a normal call, the caller would dial a destination number, and the network would complete the call as dialed. To activate the personal dial tone service, however, the subscriber dials an access number assigned to that service, such as 1-800-DIALTON, from the station 1_A (step S33).

The dialing of an outgoing call, in this case to the access number, is another type of service request. The central office switch 11_1 recognizes the dialed access number as a trigger event or 'PIC' (step S34). The SSP type central office 11_1 creates a TCAP query message containing relevant information, such as the office equipment (OE) number and/or telephone number assigned to the off-hook line, the dialed number and the type of triggering event. The office 11_1 transmits that query to the SCP 19 (step S35). Specifically, the SSP central office 11_1 transmits the query over an SS7 link to one of the STPs 15. The query includes a point code and/or a global title translation addressing the message to the SCP 19, and the STP 15 relays the query message over the appropriate link to the SCP 19.

In response to a query, the SCP 19 accesses its database to determine how to process the particular call. In this case, the query indicates the dialed number type trigger and provides the digits of the specific number dialed. The SCP 19 uses the dialed number as the access key. The SCP 19 retrieves a call processing record (CPR) corresponding to that number associated with the personal dial tone access function (step S36). For the current exemplary access, the CPR will provide information necessary for routing the call to the IP 23 that will perform the necessary speaker identification/verification (SIV).

Based on the CPR, the SCP 19 formulates a response message instructing the SSP central office 11_1 serving the customer to route the call. In this case, the message includes information, e.g. a office equipment (OE) number or telephone number, used for routing a call to the identified IP 23. The SCP 19 formulates a TCAP response in SS7 format and transmits the TCAP response message back to the SSP central office 11_1 (see step S37).

The SSP type switch in the central office 11_1 uses the routing information to connect the call to a line or channel to the IP 23. A voice grade call connection now extends between the calling station 1_A and the IP 23 (step S38).

The central office 11_1 provides a signaling message to the IP 23 with the call. In this case, the signaling message includes the dialed digits indicating a call to the personal dial tone access number. The signaling message also includes either the office equipment number or the telephone number of the line to the calling station 1_A.

As in the earlier example, the IP 23 will seize the line for the incoming call and launch a query to the SCP 19 through the TCP/IP network 27 (step S39). The SCP 19 accesses an appropriate CPR (S40), and based on that CPR, the SCP 19 transmits back a message (S41) instructing the IP 23 to execute a program or script for the dial-up access to the personal dial-tone service.

The IP initially plays a greeting and a prompt message (S42) and collects spoken input information (S43). The IP 23

may also play a prompt and collect digits representing the subscriber's normal or home telephone number. The voice authentication module 233 analyzes the spoken identification information to extract characteristic information (S44) and compares the extracted information to stored template or feature data to determine if there is an adequate match to the known subscriber data (S45), as in the earlier example.

In step S45, the IP 23 determines if the information extracted from the speech input matches any of the stored template data feature data for an identifiable subscriber. If there is a match, the IP now knows the identity of the calling subscriber. Based on the identity of the subscriber, the IP 23 obtains the subscriber's profile data from the central office 11_N serving the subscriber's home telephone line. If the IP 23 is in direct signaling communication with the home central office 11_N, for example via an ISDN D-channel or an SMDI link, the IP 23 may directly request and receive the profile data over the signaling link. If the IP and the switch are not in direct communication, the IP may provide a message notifying the SCP 19, and the SCP 19 would obtain the data from the switch and provide it back to the IP 23.

The IP 23 formulates a D-channel signaling message containing the subscriber's profile information together with an instruction to load that information into the register assigned to the call in place of the profile information corresponding to the off-hook line (step S46). The IP 23 supplies the message to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link. In response, the administrative module processor 61 rewrites the profile data in the register assigned to the call using the data from the IP 23 (step S47). As such, the profile information in the assigned register now corresponds to the identified subscriber.

When the central office switch 11_1 reloads the profile, the central office disconnects the link to the IP 23 and connects tone receivers to the caller's line. The central office 11_1 may also provide a standard dial tone or other message over the line (step S48). The caller can now dial digits in the normal manner (step S49), and the switch in the central office 11_1 will load the dialed digits into the assigned register within the call store 67. The central office 11_1 utilizes the dialed digits and the subscriber's profile data to process the call (step S50). For example, the switch in central office 11_1 may provide the appropriate record to bill the outgoing call to the subscriber's account. In accord with the invention, the network also provides the subscriber specific information for caller ID purposes, in the manner discussed in detail above relative to either FIG. 4B or FIG. 4C.

As in the earlier example, the preferred embodiment allows up to N tries or attempts to provide recognizable subscriber identification information. Thus, if in step S45 the extracted information characterizing the input speech signals did not match any of the templates or feature data used by the IP 23, then the process flows to step S51. If the current number of attempts for recognition on this call is less than N, processing returns to step S42, and the IP 23 again transmits the prompt for speech input. The caller again speaks the requested input information (S43), and the authentication module 233 again analyzes the input information (S44). If the second input adequately matches a stored subscriber's information S45, the processing flows through steps S46 to S50 to complete the call as described above.

However, if the extracted speech information does not match a stored subscriber template or feature data, processing again flows to step S51. If the number of tries now corresponds to the limit N, the processing branches to step S52. The IP 23 preferably transmits a message indicating denial of service (S52), and then transmits a message to the central office 11_1 signifying disconnection of the access call (S53). It should be noted that, in this example, normal service provided over the line to station 1_A is available on a subsequent call. The failure to recognize the caller as a personal dial tone subscriber only prevents the caller from using the personal dial tone services of a subscriber to that service, for example specialized billing of calls to that subscriber's account instead of to the account normally associated with the line to the calling station 1_A.

In the above discussed dial-up access example, the dial tone service was personalized for a single outgoing call by temporarily loading the subscriber's profile data into the register assigned to the outgoing call in the originating central office 11_1. The system can provide such service to the subscriber over any line or to any telephone station, including pay telephone stations. For this and other reasons such privilege to make outside calls is typically not granted to persons subject to home incarceration.

The present invention also enables activation of the personal dial tone service on a particular line for some predetermined period of time, for example to enable use of office or business services from some remote location while a business subscriber is out of the office. This type of operation involves an activation call requesting the service on a particular line for the desired period. Consider now an example of such a time activated service.

For this example, assume that a subscriber's normal business telephone is telephone 1_B. The end office switch 11_N stores the subscriber profile data for the line associated with that telephone station. Now assume that the subscriber is using station 1_A connected through a telephone line to central office 11_1 for business related communication services. The business related communication services include both incoming call related services and outgoing call related services.

To activate the personal dial tone service, the subscriber again lifts the handset at station 1_A, receives dial tone from the central office 11, and dials the access number assigned to that service. The network uses AIN type processing to route the call to the IP 23, as in the example discussed above relative to FIG. 5.

As in the earlier examples, the IP 23 seizes the line for the incoming call and launches a query to the SCP 19 through the TCP/IP network 27. The SCP 19 transmits back a message instructing the IP 23 to play a greeting and a prompt message and collect and analyze spoken input information to identify and authenticate the subscriber. The instruction from the SCP 19 also causes the IP 23 to prompt the subscriber and obtain input information regarding the time period for service activation and possibly to obtain digits representing the subscriber's normal business telephone number. The process of calling the access number and interacting with the IP to activate the personal dial tone service on a line for the desired period is another type of service request.

For outgoing call processing, the IP 23 signals the central office 11_1 serving the line to station 1_A to set an off-hook trigger in the profile data associated with that line. The IP also obtains the profile information from the switch 11_N serving the station 1_B and provides that information together with a virtual OE number to the central office 11_1. The office 11_1 stores the profile in its disc memory 63 in such a manner that the switch in that office can use the virtual OE number

to retrieve that subscriber's profile. For incoming calls to the subscriber, the IP 23 transmits a signaling message to the subscriber's home office 11_N to set a terminating attempt trigger (TAT) against the line to the subscriber's office telephone 1_B.

The IP 23 also transmits a message through the TCP/IP network 27 to the SCP 19 advising the SCP 19 of the service activation. This message identifies the subscriber, for example by their normal telephone number and identifies the telephone number and office equipment (OE) number associated with the line to station 1_A that the subscriber selected for their personal dial tone service.

In response to the message from the IP 23, the SCP 19 now establishes or modifies two CPRs for this subscriber. One CPR controls processing of calls to the subscriber's normal business telephone number to enable routing to the station 1_A, and the other controls routing of outgoing calls from that station to the IP 23 for speaker identification/verification (SIV) processing.

Subsequently, when there is an outgoing call from the station 1_A, the network will route the call to the IP 23 to determine if the caller is the subscriber or some other party, exactly as discussed in the per-call service from a shared use line (FIG. 4). As in that earlier example, if the IP identifies the caller as the personal dial tone subscriber, then the IP 23 provides the virtual OE number to enable loading of subscriber's profile from disc memory 63. The network provides the telephone number and the subscriber specific information, for caller ID purposes, as discussed above. If the IP determines that the caller is not the personal dial tone subscriber, the IP instructs the originating office 11_1 to simply provide dial tone and complete the call in the normal manner. The central office 11_1 therefore will utilize the office equipment (OE) number and profile information normally associated with the line, instead of those for the personal dial tone subscriber. The network provides caller ID service, identifying the number and possibly the main name associated with the line, in the normal manner. In this way, it is quite easy for the personal dial tone subscriber and the normal subscriber to both obtain their desired services on their respective calls via the same line, and to be correctly identified to called parties who subscribe to caller ID services.

The trigger set against the subscriber's normal telephone number and establishment of the CPR in the SCP 19 enables redirection of calls normally intended for the subscriber's business telephone 1_B to the line to station 1_A. Depending on how the subscriber elects to define their individual service, the network may simply route the calls to the line to station 1_A, as a normal AIN forwarded call that simply rings the station(s) 1_A on the line. Alternatively, the subscriber may elect an enhanced service which involves routing to the IP, IP prompting and speech recognition to identify the called subscriber and distinctive ringing over the line, in a manner analogous to that used for processing incoming calls in shared use applications, such as the above discussed dormitory example.

As noted above, the dial-up access procedure in this latest service example required the subscriber to specify a time period that the personal dial tone service should apply to the particular line. The IP 23 stores a record of the time period elected by the subscriber. When the period expires or if the subscriber calls in earlier to change the service to another line or temporarily cancel the service, the IP 23 will provide cancellation notices to the appropriate central offices 11 and to the SCP 19. In the example, the IP 23 will notify the office 11_1 to cancel the off-hook trigger set against the line to station 1_A and to delete the subscriber's virtual OE number and profile from its disc memory. The IP 23 will also instruct the central office 11_N to cancel the terminating attempt trigger set against the subscriber's business line to station 1_B. The notice to the SCP 19 causes the SCP to deactivate the personal dial tone CPR and the call redirection CPR. If the associated personal identification functionality for caller ID service relies on a central database, such as LIDB, the IP would also instruct that database to temporarily establish a subscriber account record associating the subscriber's name and calling card billing information with the telephone number and a subscriber identifier code.

The subscriber can then or later interact with the IP 23 to establish time based temporary personal dial tone service through another line or location, as discussed above. In this manner, a subscriber might set up a temporary office in a motel in one city for several days. The subscriber might cancel the service while in transit to a new location. Then the subscriber might reestablish the service to set up a temporary office service at a vacation home for a week.

The time based personal dial tone service could be modified in several manners. For example, the subscriber might establish a file for use by the SCP or the IP to establish the personal dial tone service at two or more locations at specified times, e.g. at the office during office hours and at a home office during other hours. Also, the above example of this service relied on downloading the subscriber's profile into the switch serving the line with which the subscriber is temporarily associated. Alternatively, the IP could obtain the profile from the subscriber's home switch and provide the profile to the serving switch as part of the processing of each outgoing call by the subscriber from that line during the specified time period.

A preferred network implementation and a number of specific call processing routines have been discussed above by way of examples relating to the present invention. However, the preferred embodiment of the invention is amenable to a variety of modifications.

Also, the currently preferred embodiment utilizes AIN routing to the IP and speaker identification/verification elements within the IP to identify the subscriber for profile selection. As speaker identification/verification equipment becomes more readily available, cheaper and more compact, it will be possible to build this functionality into the line cards of the end office switches. The switch itself will challenge the caller, analyze spoken information and identify the subscriber to select the appropriate profile, without routing to an IP or the like.

The foregoing description sets forth the arrangement and functionality of the enhanced telecommunications network of the invention which enable an improved form of home incarceration. Following is a description of an illustrative example of a preferred mode of operation of that system. Referring to FIG. 6 there is shown an amplified version of the network of FIG. 1 wherein like reference numerals refer to like elements. In the arrangement of FIG. 6 the telephone terminals 1 and 1_A are located on the premises of an incarceration site 2. The site may constitute the detainee's home, a governmental institution, a half-way house, or the like. In many instances home incarceration services are provided by governmental agencies of a state or county. Alternatively, such services may be provided by private companies for the state or county, as the case may be.

In either event the entity providing the service typically utilizes a supervisory node or incarceration control center,

such as that which is shown at 3 in FIG. 6. Such a control center would normally contain a series of telephone terminals for contacting and receiving calls from detainees. Such telephones are represented by the terminal 7 in FIG. 6. The center would also contain a control processor 8 and associated disk storage and database 9. It will be appreciated that at least most of the routine verification calls made and received by the control center are automated in known fashion. The telephone terminal and processor are connected to the central office 11, preferably by an ISDN link, and connected to the IP 23 by a data link.

Assuming that the incarceration site 2 houses four detainees A, B, C, and D, one illustrative example of operation of the system may be described as follows.

The incarcerating authority has established control parameters for each detainee which define the number and frequency of incarceration verification checks to be made daily to insure compliance with the program. These parameters are imparted to the control center 3 and the necessary programming of the control center and telephone facilities is accomplished. In this example it is assumed that the control parameters require verification checks of detainee A a minimum of three times per day with an interval between checks not to exceed eight hours.

During a calendar day when detainee A makes no outgoing telephone calls, the control center 3 will initiate three calls to A commencing at approximately 7 AM. Calls will thus be initiated at approximately 7 AM, 3 PM, and 11 PM. Each of these calls may be made to the telephone number of one of the lines to the telephone terminals 1 and 1_A on the incarceration premises.

On an incoming call the network prompts the caller to speak the name of the called party. The network would then recognize the spoken name, the necessary signaling occurs between the telephone, central office switch, signal control point and intelligent peripheral, and the switching office loads the appropriate profile, as described in detail hereinbefore in connection with FIG. 4. The distinctive ringing signal appropriate to the called detainee is sent to the line.

In the case where a verification call is made by the control center the identity of the appropriate profile is known and signaling between the control center and network may replace the voice recognition steps to obtain the desired individualized distinctive ringing signal. Alternatively, the control point may respond to the name query or the prompt by transmitting a prerecorded voice signal of the called detainee.

Assuming the detainee to whom the ringing signal has been assigned responds and picks up the telephone, he/she will be directed to call back the control center. In the return call the calling detainee is directed to speak his/her name and perhaps additional specified words for which the IP has previously prepared templates for the particular detainee. Upon voice verification by the IP the calling detainee may receive a predetermined prompt, such as "Thank you. Goodbye." The call is then terminated.

The network date and time stamps the successful incarceration verification or validation and may make and store a recording of all or part of the dialog. Pertinent data regarding the exchange is periodically reported from the network to the control center. The control center also date and time stamps the call and can correllate the two sets of records to create a complete history of the event. Both the telephone network and the incarcerator control center may utilize the collected data for billing purposes. The Telco may bill the entity operating the control center, and that entity may bill the county or state. Alternatively, the county or state may operate the control center wherein only one billing would be necessary.

In the foregoing example it has been assumed that the called line was available and was not busy. In the event that the line is busy the control center may simply replace the call to the other line to the incarceration site. If both lines are busy the control center can contact the intelligent peripheral over its data link. The intelligent peripheral knows which detainee is a party to the call, and the control center can obtain this information from the IP. In addition, all or a portion of the call may be monitored by the IP. The control center may obtain access to the recording for further verification or downloading if desired. In this manner the system of the invention makes it possible to utilize on-going third party calls for verification or validation purposes. If the presence of the desired detainee can be validated in such a manner it becomes unnecessary for the control center to further pursue that particular validation, thereby simplifying and economizing the incarceration process.

According to another feature of the invention the IP may store a record of the date, time, and line on which a successful match of the name of a detainee has occurred. This information may then be further processed in the IP or forwarded to the control center on a periodic basis. This data may then be utilized to verify periods of time within which the verification requirements of a particular detainee have been satisfied. Such a procedure makes it possible to obviate the necessity of the incarcerating supervising entity to initiate calls to verify incarceration.

If the program for a specific detainee requires verification at least once every eight hours, the occurrence of verification during a third party call may be utilized to shift back the time for the next verification by call from the control center. In this way it is possible to greatly reduce the number of calls from the control center which are required within a sequential twenty-four hour period. The control center receiving a signal that a verification was confirmed at a specific time can slide the scheduled time for the next required call back to take advantage of the validation.

In the event that the detainee which the control center seeks is not a party to either of the ongoing calls on the two incarceration site lines, the control office may interrupt one or both of the calls. The control center may then transmit to the interrupted callers a prearranged signal directing that the call be terminated within a specified amount of time, such as two or three minutes. The termination of the call will free the affected line and the call from the control center to the detainee may then be replaced and com pleted.

In the foregoing example it has been assumed that the called detainee satisfied the voice authentication procedure. In the event that the name and possible other key word(s) do not find a matching template in the IP on the initial attempt, the IP may prompt for one or more repeat attempts by the called party. If the voice verification is not accomplished in a pre-established maximum number of attempts, the IP may notify the party that the test failed and give default instructions. The IP also provides a failure signal to the control center and activate whatever procedure that the county or state has mandated for such situations.

In typical home incarceration circumstances the detainees would be permitted some limited use of the telephone for personal purposes. Thus a particular detainee might be permitted any local calls but long distance calls may be prohibited. In another case the detainee may be restricted to local calls to a limited number of telephone directory

numbers. The system of the invention provides a ready implementation of such restrictions.

Operation is now described in the situation where an originating call is placed by a detainee from the incarceration site.

The detainee lifts the handset creating an off-hook state in the telephone and a corresponding signal or change in state on the line to the central office. In this call flow, the off-hook signal is a service request to make an outgoing call. The serving central office detects the off-hook and commences its call processing. Specifically, the central office assigns a register to this call in the call store 67 and loads profile information associated with the off-hook line from the disc storage 63 into the assigned register. In this case, the central office is an SSP capable office, and the loaded profile data indicates an off-hook immediate trigger set against the particular line. The serving SSP type office therefore detects this off-hook PIC as an AIN trigger.

In response to the off-hook and the off-hook trigger set in the profile, the SSP type central office switch launches a query to the SCP 19. Specifically, the SSP creates a TCAP query message containing relevant information, such as the office equipment (OE) number assigned to the off-hook line, and transmits that query over an SS7 link to one of the STPs 15. The query includes a destination point code and/or a global title translation addressing the message to the SCP 19, and the STP 15 relays the query message over the appropriate link to the SCP 19. The query from the SSP central office identifies the caller's line by its associated office equipment (OE) number.

In response to a query, the SCP 19 accesses its a database, typically, the MSAP database set up in the ISCP, to determine how to process the particular call. The SCP 19 identifies an access key in the query and uses the key to retrieve the appropriate record from the database. In this case, the query indicates an off-hook trigger as the trigger event, therefore the SCP 19 uses the calling party office equipment (OE) number as the access key. The SCP 19 retrieves a call processing record (CPR) corresponding to the office equipment (OE) number associated with the off-hook line and proceeds in accord with that CPR.

For the present example of the personal dial tone service, the CPR will provide information necessary for routing the call to some node of the network that will perform speaker identification/verification (SIV), in this example the IP. Therefore the CPR provides information for routing the call to the nearest available IP having the SIV capability.

Based on the CPR, the SCP 19 formulates a response message instructing the SSP central office serving the line to route the call. In this case, the message includes information, e.g. a office equipment (OE) number or telephone number, used for routing a call to the identified IP 23. The SCP 19 formulates a TCAP message in SS7 format, with the destination point code identifying the SSP office. The SCP 19 transmits the TCAP response message back over the SS7 link to the STP 15, and the STP 15 in turn routes the TCAP message to the SSP central office.

The SSP type switch in the central office 11, uses the routing information to connect the call to one of the lines or channels to the IP 23. A two-way voice grade call connection now extends between the calling station and the IP 23. In the present example, the switch actually connects the off-hook line to the line to the IP before providing dial tone to the caller.

As noted above, the communication link to the IP 23 provides both line connections and signaling, preferably over a primary rate interface (PRI) type ISDN link. When the central office extends the call from the calling party's line to a line circuit (over a B channel) to the IP 23, the switch in that office also provides call related data over the signaling link (D channel for ISDN). The call related data, for example, includes the office equipment (OE) number normally associated with the off-hook line and possibly the telephone number for that line.

In response to the incoming call, the IP 23 will seize the line, and it will launch its own query to the SCP 19. In the preferred network illustrated in FIG. 1, the IP 23 and the SCP 19 communicate with each other via a separate second signalling network 27, for example utilizing either an 1129+ protocol or a generic data interface (GDI). The query from the IP 23 again identifies the caller's line by at least its associated office equipment (OE) number.

In response to the query from the IP 23, the SCP 19 again accesses the appropriate CPR and provides a responsive instruction back through the network 27 to the IP 23. Although the IP 23 could passively monitor any speech that the user might utter, the preferred implementation utilizes a 'Challenge Phase' to prompt the detainee to input specific identifying information. In this case, the instruction causes the IP 23 to provide a prompt message over the connection to the caller. Here, the signal to the caller may be a standard dial tone or any other appropriate audio signal. Preferably, the instruction from the SCP 19 causes the IP 23 to provide an audio announcement directing the detainee caller to speak personal information. In one preferred example, the IP plays an audio prompt message asking the caller, 'Please say your full name'. The process may ask for any appropriate key word or phrase identifying information.

The signal received by the IP 23 goes over the lines and through the central office switch(es) for presentation via the off-hook telephone to the calling party (detainee). In response, the caller will speak the identifying information into their off-hook telephone, and the network will transport the audio signal to the IP 23.

When the IP 23 receives speech input information during actual call processing, for this service example, the IP analyzes the speech to extract certain characteristic information.

The IP 23 stores a template or other voice pattern information for each person who has the personal dial tone service in the area that the IP normally services. This would normally include the calling detainee. If the IP 23 does not store the particular template or feature information it needs to process a call, the IP 23 can communicate with a remote IP 23_R to obtain that information. In the present shared line example, the IP 23 will store template or feature data for each subscriber or detainee associated with the particular off-hook line.

When the IP 23 receives input speech and extracts the characteristic information during actual call processing, the IP compares the extracted speech information to stored pattern information, to identity and authenticate the particular caller. In the present example, the voice authentication module 233 in the IP 23 compares the extracted speech information to the stored template or feature data for each subscriber associated with the particular off-hook line.

The IP 23 determines if the information extracted from the speech input matches any of the stored template data feature data for an identifiable subscriber or detainee (within some threshold level of certainty). If there is a match, the IP now knows the identity of the calling subscriber or detainee. Based on the identification of the calling subscriber or

detainee, the IP 23 selects a virtual office equipment (OE) number from storage that corresponds to the subscriber or detainee.

The IP 23 formulates a D-channel signaling message containing the virtual office equipment (OE) number together with an instruction to load that OE number into the register assigned to the call in place of the OE number of the off-hook line. The IP 23 supplies the message to the SSP central office switch 11_1 over the D-channel of the ISDN PRI link. In response, the administrative module processor 61 rewrites the OE number in the register assigned to the call using the OE number received from the IP 23.

Upon rewriting the OE number in the register, the administrative module processor 61 of central office switch 11_1 also reloads the profile information in the register. Specifically, the administrative module processor 61 retrieves profile information associated with the virtual office equipment (OE) number from the disc storage 63 into the register. As such, the profile information in the assigned register in the call store 67 now corresponds to the identified subscriber or detainee, rather than to the off-hook line.

The profile information provides a wide range of data relating to the subscriber or detainee's services. The profile data provides necessary billing information, enabling billing from the call to this particular subscriber or detainee for a personal call and/or billing to the incarceration supervision entity. The profile also defines various service features and restrictions applicable to this subscriber or detainee on outgoing calls, such as three-way calling in the case of an allowed service, and prohibited and/or allowable calls in the case of a detainee. The profile may also include a list of key words for which monitoring is maintained during calls. The profile may define a class of calling service available to the subscriber or detainee. Long distance calls are usually proscribed in the case of detainees. To this end the key word list may contain credit card names and DTMF sequences which precede long distance calls. The profile may also direct that that call be completely monitored or stored.

In the presently preferred implementation, when the central office switch 11_1 reloads the profile, the central office disconnects the link to the IP 23 and connects tone receivers to the caller's line. Optionally, the central office may provide a 'dial tone' or other message over the line. The caller now dials digits in the normal manner, and the switch in the central office 11_1 loads the dialed digits into the assigned register within the call store 67. The central office 11_1 utilizes the dialed digits and the subscriber's profile data to process the call. For example, if the dialed digits represent a call within the subscriber's permitted class of service, the switch completes the call to the destination station using the dialed digits in the normal manner. If the profile data requires a particular billing treatment, e.g. to bill a long distance call to the subscriber, the switch makes the appropriate record and forwards the record to the exchange carrier company's accounting office equipment. In accord with another aspect of the invention, the network provides caller ID data naming the identified subscriber or detainee to the destination station.

The processing to complete the call involves a sequence of steps. Of particular note, some of these steps facilitate delivery of caller ID information to the destination station. The present invention involves delivering caller ID information which corresponds to the identified subscriber or detainee, preferably the subscriber or detainee's name, rather than simply the number of the line or station from which the subscriber or detainee initiates the call.

While the foregoing has described what are considered to be preferred embodiments of the invention, it is understood that various modifications may be made therein and that the invention may be implemented in various forms and embodiments, and that it may be applied in numerous applications, only some of which have been described herein. It is intended by the following claims to claim all such modifications and variations which fall within the true scope of the invention.

What is claimed is:

1. A method comprising:

for a designated person, generating service profile data for controlling a sequence of operation in a multiple link communication network relating to provision of one or more communication services through the network;

storing the service profile data for the designated person with other service profile data in the communication network;

detecting a request to make a call from a predetermined link of the communication network;

receiving and processing speech signals from a person making said request via the predetermined link to identify the requesting person as said designated person;

instructing a switching office of the communication network to utilize the service profile data for the designated person for processing of the call from the predetermined link through the network; and

storing data regarding the identification of said designated person and data regarding said call.

2. A method as in claim 1, wherein the step of instructing the switching office comprises providing a virtual office equipment number for use in retrieving the service profile data for the designated person.

3. A method as in claim 1, wherein the step of receiving and processing speech signals comprises comparing information characteristic of the received speech signals to stored speech characteristic data corresponding to a plurality of subscribers.

4. A method as in claim 1, wherein the communication network comprises a telephone network having a plurality of central offices each of which serves a plurality of communication links.

5. A method as in claim 4, wherein the plurality of communication links comprise telephone lines.

6. A method as in claim 1, including the step of transmitting at least a portion of said stored data to a control node.

7. A method as in claim 1, wherein said stored data includes the identity of said designated person, the date and time and duration of the call.

8. A method as in claim 1, wherein the stored data regarding the identification comprises the approximate time of the identification.

9. A method as in claim 8, wherein the stored data regarding the identification further comprises the identity of said designated person.

10. A method as in claim 9, wherein the stored data regarding the identification provides a verification of home incarceration.

11. A method, comprising:

identifying one party to a requested communication service as one of a plurality of designated persons;

using a virtual office equipment number, assigned to the identified one person, to retrieve corresponding profile data from stored profile data for the plurality of designated persons;

providing communication service over a link of a communication network based at least in part on the retrieved profile data; and

using a portion of the retrieved profile data to provide an identification of the one person over another link of the communication network.

12. A method as in claim 11, wherein the step of identifying one person comprises analyzing speech information from the one person.

13. A method as in claim 11, wherein the communication service comprises attempting to complete a call from the one identified person through the communication network to said another link, and a terminal coupled to said another link presents the identification of the one person to a called party.

14. A method as in claim 13, wherein the identification of the one person comprises the name of the one person.

15. A method as in claim 14, wherein the step of using data to provide the identification comprises transmitting data representing the name of the one person from the profile data through the communication network to said terminal.

16. A method as in claim 14, wherein the step of using data to provide the identification comprises:

transmitting data representing the identity of the one person to a database;

accessing information in the database to translate the data representing the identity of the one person into data representing the name of the one person; and

transmitting the data representing the name of the one person to the terminal.

17. A method as in claim 11, wherein:

the communication network comprises a telephone network having a telephone switching system; and

the step of using the virtual office equipment number comprises supplying the virtual office equipment number to the telephone switching system and retrieving the corresponding profile data from storage in the telephone switching system.

18. A method as in claim 11, further comprising the step of creating a record comprising data regarding the identification of the one person.

19. A method as in claim 18, wherein the data regarding the identification of the one person comprises the approximate time of the identification.

20. A method as in claim 19, wherein the data regarding the identification of the one person further comprises the identity of the one person.

21. A method as in claim 20, further comprising verifying home incarceration using the record.

22. A method as in claim 18, wherein the record further comprises data regarding the communication service provided.

23. A telecommunication network comprising:

a central office for processing calls originated over a plurality of communication links, said central office including mass storage containing service profiles of designated users of said communication links; and

a peripheral coupled to the central office, said peripheral including a voice authentication module for analyzing speech of a caller from one communication link to identify the caller as a designated person and provide an office equipment number assigned to the identified designated person to the central office, wherein:

the central office retrieves a service profile corresponding to the office equipment number from the mass storage and processes at least one call over a communication link using the retrieved service profile, and

the network produces a record, comprising data regarding the identification of the caller as the identified designated person and data regarding said call.

24. A network as in claim 23, wherein the the office equipment number comprises a virtual office equipment number corresponding to the caller.

25. A network as in claim 23, wherein the central office comprises a first telephone switch, and the network further comprises a second telephone switch interconnected to the first telephone switch via a trunk circuit.

26. A network as in claim 25, wherein at least some of the communication links comprise telephone lines.

27. A network as in claim 25, further comprising:

a service control point, remote from the telephone switches, said service control point containing a database of call processing records; and

a first signaling network separate from the communication links and trunk circuits for carrying signaling messages between the service control point and the telephone switches.

28. A network as in claim 27, wherein the peripheral includes a communication server facilitating signaling communication between the peripheral and the service control point.

29. A network as in claim 28, further comprising a second signaling network, separate from the communication links, the trunk circuits and the first signaling network coupled between the communication server and the service control point.

30. A network as in claim 29, wherein the peripheral stores characteristic voice feature data relating to a plurality of designated persons, and the peripheral comprises a voice authentication module for comparing characteristic data extracted from speech of the caller to the stored characteristic voice feature data.

31. A network as in claim 30, further comprising a remote peripheral storing characteristic voice feature data relating to another plurality of designated persons, wherein the remote peripheral is coupled to the second signaling network for communication of stored characteristic voice feature data.

32. A network as in claim 29, wherein:

the first signaling network comprises a common channel interoffice signaling network coupled to the telephone switches; and

the second signaling network comprises a packet switched data network.

33. A network as in claim 32, wherein:

the common channel interoffice signaling network utilizes signaling system 7 (SS7) protocol; and

the packet switched data network utilizes transmission control protocol/Internet protocol (TCP/IP).

34. A method comprising:

for a designated person, generating service profile data for controlling a sequence of operation in a multiple link communication network relating to provision of one or more communication services through the network;

storing the service profile data for the designated person with other service profile data in the communication network;

detecting a request to make a call from a predetermined link of the communication network;

receiving and processing speech signals from a person making said request via the predetermined link to identify the requesting person as said designated person;